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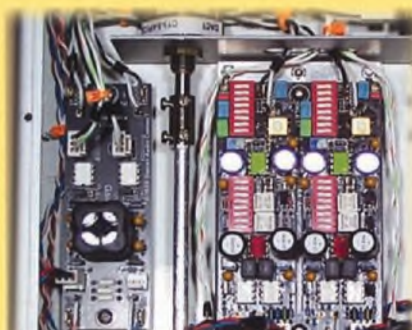

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# Guest Editorial

## Remembering Digital's Father

**By Reg Williamson**

Sixty-three years ago saw the most significant development in audio engineering since Edison's work on the "phonograph" in the late 19th century—the invention of pulse-code modulation, or PCM; and the man behind it was Alec Harley Reeves.

### BACKGROUND

Reeves was born in Redhill, Surrey, in 1902. He was, perhaps, everyone's concept of the eccentric inventor, undeviating in his devotion to his science. His "hibernations" when he disappeared into his laboratory for days at a time were legendary, apparently not bothering to go home until some particular problem or other had been solved.

Yet, despite this, he had the unique ability to put across his ideas in a language perfectly understandable by the layman. Those who were fortunate enough to hear one of his lucid and witty lectures on the future pattern of telecommunications were impressed as much by this ability as by his brilliant mind. His eccentricity extended to an interest in parapsychology, and he actually put together experimental apparatus for attempted direct pickup of his thoughts.

Reeves rarely came across as one of the outstanding scientists of his day, being a gentle and modest man and spending much of his private life helping others, including youth and community work. He never married, but had a legion of friends, and yet remained essentially a lonely man. He lived for many years with his sister, a retired schoolmistress. After her permanent confinement in the hospital, he seemed to withdraw into himself, but his scientific mind remained as keen as ever.

### DISTINGUISHED CAREER

He began his career in the French laboratories of the giant American company ITT, and played a major role in establishing the first radio-telephone link across the Atlantic in the mid-20s. Over the next ten years, his main work was on shortwave and microwave radio, then very much in its infancy. But he made his most significant breakthrough in the mid-1930s.

Reeves had been trying to find a solution to the problems of noise, distortion, and crosstalk that multi-channel radio links often suffered. He proposed a system in which voice signals were electronically coded into streams of pulses before transmission and then converted back into analog sound at the receiving end.

This was Pulse Code Modulation, or what we now use as a generic title, digital encoding. His basic patents, filed on October 3, 1938, in France, also included proposals for error correction, a fundamental prerequisite for all forms of digital transmission and recording. Patents were also filed in the US a year later, on November 22, 1939. Reeves escaped to the UK in 1940 after the occupation of France in World War II, returning to make a major contribution to the war effort with numerous inventions.

By 1943 a basic PCM system using tubes was in use for secret long-distance radio telephone links, such as that from Churchill's wartime bunker in England to President Roosevelt. However, the equipment was so bulky and cumbersome that most of it had to be located in the basement of a large department store in London. Reeves was reluctant to work on offensive weapons, but his inventions included the multipoint gas counting tube; the forerunner of Radar, a radio navigation system called "Oboe," and probably the war's most accurate bomber guidance technique. For this, he was awarded the MBE in 1945.

### LEGACY

It was the arrival of the ideal electronic switch, the transistor, that spurred along the commercial evolution of digital techniques for telephony. In November 1962, the first digital telephone junction network was inaugurated between two central offices in London, England. Even then, the system was only cost-effective for routes up to 30 miles long, so development was slow and confined to enlarging the capacity of existing orthodox cable links.

For the (then) British General Post Office Engineering Dept., these benefits were a greater attraction than the much improved fidelity of audio transmission, with the maintenance of audio quality now totally independent of the signal path length. In recent years, it was the emergence of mass-produced integrated circuits that made possible the full digital working of the entire telephone system—even to the customer's telephone which is with us today.

For high-quality transmission of sound, credit must go to the Research Dept. of the BBC. By 1972, they had begun a policy of linking their studios to their TV/sound and FM transmitters with a digital network, initially with a "Sound-in-Synch" system of incorporating the TV sound in the horizontal synch pulse of the video waveform. Then, their VHF stereo sound feeds involved a linear-encoded 13-bit system to all their FM transmitters throughout

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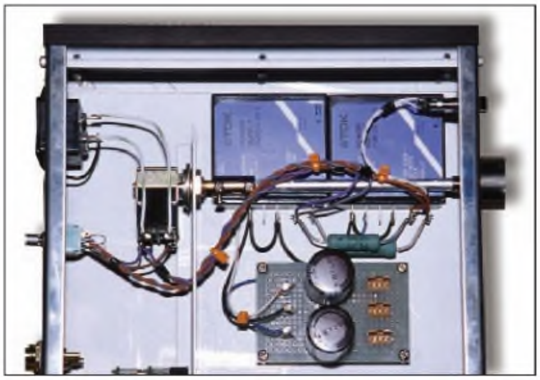

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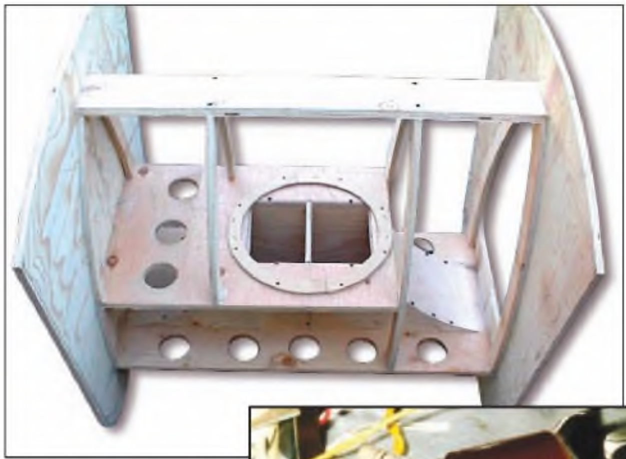

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*The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.*

JOHN STUART MILL


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## Guest Editorial

*from page 2*

the UK. This program distribution network is even now unique to the UK.

While the BBC did some important experimental work in digital recording, this application of PCM was developed more fully by NHK, the state broadcasting company, and Nippon Columbia in Japan. Over a period of three days, from April 24–26 in 1972, Nippon Columbia made the world's first commercial digital recording in Aoyama Hall in Tokyo, using a modified standard quad video recorder. The sampling rate was 47.25kHz with 13 bits per linear sample. It was of the Smetana Quartet playing Mozart and was subsequently released on LP in October of the same year under the Denon label. The same historic recording is now available on CD.

Most of Dr. Reeves' richly deserved honors came late in life, particularly when the significance of PCM became clear. In the US, the Franklin Institute of Philadelphia presented him with the prestigious Ballantine Medal in 1965 for his invention. In 1969, the Queen awarded him the CBE for his contribution to telecommunications. In a rare gesture in his honor, the British Post Office issued a special postage stamp on the subject of PCM. And in that same year, the first pictures of Mars were sent back to Earth by Mariner 4 using a PCM signal.

Active until the end and working on optical fiber techniques, he received an honorary doctorate from the University of Essex, England, just four months before his death from cancer at 69 on October 13, 1971. There are over 100 patents to his credit; but above all, no one should minimize his remarkable contribution to the science of sound transmission and recording. Other digital formats have since been developed, but his original pulse-code modulation technique is still at the heart of most commercial formats and will remain so.

So, when you next put on your CD, remember this quiet, modest man who made it all possible. ❖

# An Easy Solid State Preamplifier

This amplifier provides for easy assembly with Danish-made module amplifiers and peripheral components. This article was originally published in Japan's premier high-end tube magazine, *MJ Audio Technology*, May 2000. **By Satoru Kobayashi**

This preamplifier circuit was simple to make since I chose DACT components, for which ads appear in *audioXpress*. DACT (Danish Audio ConnecT) is based in Denmark, although the sales office is somehow located in Bangkok, Thailand. It offers several unique high-end audio components such as the EQ Phono amplifier module, buffer amplifier modules, rotary switches, volume controls, stainless-steel milled knobs, and so on. Thus, DACT, which also handles Scan-Speak speakers, is the total-solution component manufacturer for high-end audio.

While accessing DACT's web pages (http://www.dact.com/), I decided I could easily build a superior preamplifier using its components and offer it to beginners who wish to construct their own amplifiers. I believed this would be as good as manufactured amps, since the particular products are nearly equal to them in quality.

Collecting the parts was very easy because DACT offers the major components I needed for this project through the Internet. I got the other parts from San-Ei Musen (now closed) in Akihabara, Tokyo. I custom-designed the case myself, and had it made by San-Ei Musen, which used stainless steel, thus giving it the appearance of a manufactured amplifier.

## CIRCUIT

Because of company policy, I assume, DACT keeps the circuit of modules itself a secret. The key component on the module was paint-shielded, so I could not see any top marking to identify its part number or manufacturer. All I can tell is that both modules use an operational amplifier. Thus I hesitate to explain the details of the circuit. But I will briefly describe the components to avoid errors in assembling them.

The major components of this amplifier (*Fig. 1*) are an RIAA equalizing amplifier (CT-100), input selector, volume control, buffer amplifier (CT-101), and the power supply. For satisfactory completion, the preamp might need some features such as tape in/out and tone control, but I removed them all to simplify assembly, since high-end audio might not need them. Please note that DACT's web page provides an Adobe pdf formatted specification and installation manual. I strongly suggest that you download the file and read it before building this project.

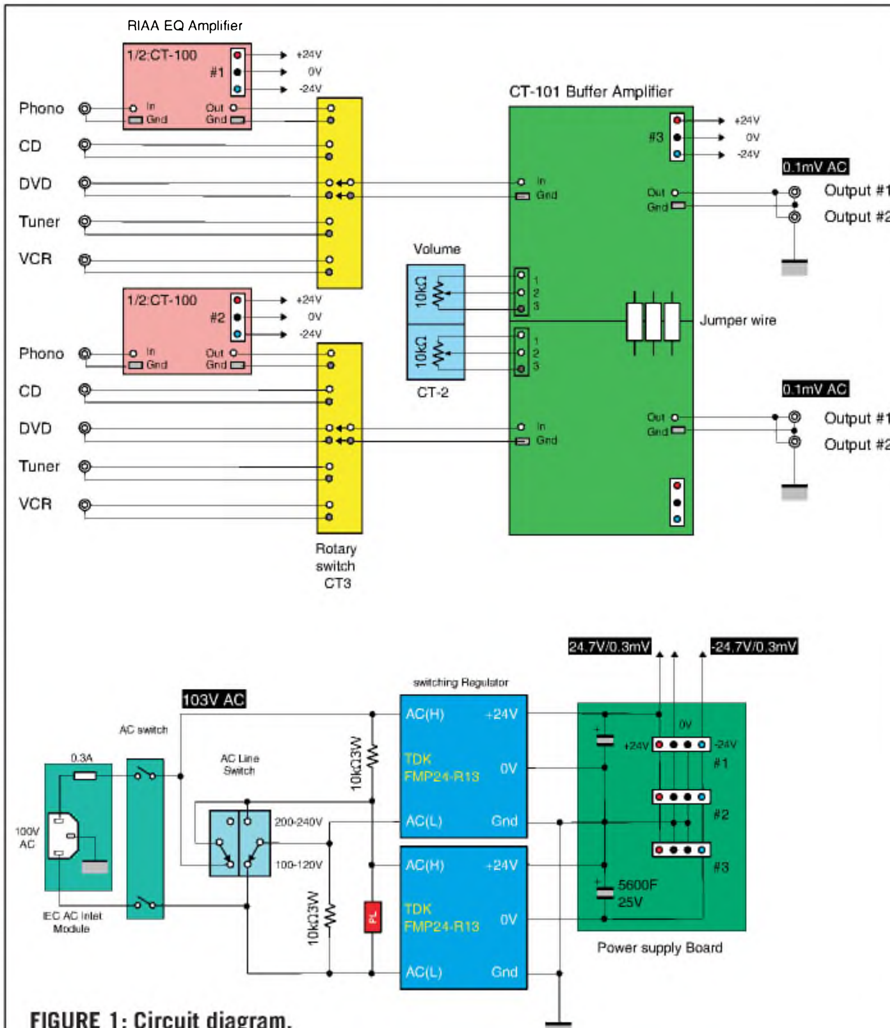
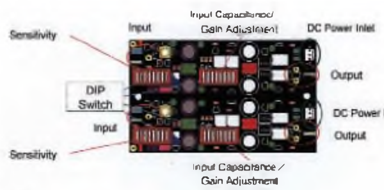



**FIGURE 1: Circuit diagram.**

The equalizing amp (CT-100) provides a feature that can make one of a number of selections, such as 10Ω, 30Ω, or 47kΩ, for input impedance, input capacitance, and time constant through DIP switches on the board (*Fig. 2*). The CT-100 even handles a 0.1 to 0.3mV signal from a moving-coil cartridge with very low impedance such as 10Ω, which obviates a step-up transformer and thus saves money. The amplifier generates up to 7V RMS at the maximum input level of 100mV with a high signal-to-noise (S/N) ratio of 98dB.

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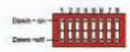

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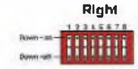


Input Impedance

| Impedance | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| 10Ω | on | on | on | on | on | on | | |
| 15 | on | off | on | on | off | on | | |
| 28 | on | off | off | on | off | off | | |
| 20 | on | off | off | off | off | off | | |
| 25 | off | on | on | on | on | on | | |
| 30 | off | on | on | off | on | on | | |
| 40 | off | on | off | on | off | on | | |
| 50 | off | on | off | off | off | off | | |
| 60 | off | off | on | on | on | on | | |
| 70 | off | off | on | on | off | off | | |
| 80 | off | off | on | off | on | on | | |
| 90 | off | off | on | off | off | on | | |
| 100 | off | off | on | off | off | off | | |
| 150 | off | off | off | on | on | on | | |
| 180 | off | off | off | on | on | off | | |
| 200 | off | off | off | on | off | on | | |
| 250 | off | off | off | on | off | off | | |
| 400 | off | off | off | off | on | on | | |
| 600 | off | off | off | off | on | off | | |
| 1k | off | off | off | off | off | on | | |
| 47k | off | off | off | off | off | off | | |

| Input C | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| 100pF | | | | | | | off | off |
| 200pF | | | | | | | on | off |
| 300pF | | | | | | | off | on |
| 400pF | | | | | | | on | on |

Gain Adjust.

| Sensitivity | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| 0.10mV | off | off | off | off | on | on | | |
| 0.12mV | off | off | off | off | off | on | | |
| 0.15mV | on | off | off | off | on | on | | |
| 0.18mV | on | off | off | off | off | on | | |
| 0.20mV | off | on | off | off | on | on | | |
| 0.25mV | off | on | off | off | off | on | | |
| 0.30mV | on | on | off | off | off | on | | |
| 0.40mV | off | off | on | off | on | on | | |
| 0.45mV | on | off | on | off | on | on | | |
| 0.50mV | off | off | on | off | off | on | | |
| 0.55mV | on | on | on | off | on | on | | |
| 0.60mV | off | on | off | off | off | on | | |
| 0.70mV | on | on | on | off | off | on | | |
| 0.80mV | off | off | off | on | on | on | | |
| 0.90mV | off | on | off | on | on | on | | |
| 1.0mV | off | off | off | on | off | on | | |
| 1.1mV | off | off | on | on | on | on | | |
| 1.2mV | off | on | on | on | on | on | | |
| 1.3mV | off | off | on | on | off | on | | |
| 1.4mV | on | off | on | on | off | on | | |
| 1.5mV | off | on | on | on | off | on | | |
| 1.6mV | off | on | on | off | on | off | | |
| 1.8mV | on | on | on | off | on | off | | |
| 2.0mV | off | on | off | off | off | off | | |
| 2.5mV | off | off | off | on | on | off | | |
| 3.0mV | on | on | off | on | on | off | | |
| 3.5mV | off | off | on | on | on | off | | |
| 4.0mV | on | on | on | on | on | off | | |
| 4.5mV | off | on | on | off | off | off | | |
| 5.0mV | on | on | on | off | off | off | | |
| 7.0mV | off | off | off | on | off | off | | |
| 8.0mV | on | on | off | on | off | off | | |
| 9.0mV | off | off | on | on | off | off | | |
| 10.0mV | on | on | on | on | off | off | | |

| Time Const. | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| 3.18off/7950off | | | | | | | on | on |
| 3.18off/7950on | | | | | | | on | off |
| 3.18on/7950off | | | | | | | off | on |
| 3.18on/7950on | | | | | | | off | off |

**FIGURE 2: CT-100 DIP-switch setup.**

## INPUT SELECTOR

CT-3, a 4-pole, 5-position rotary switch (*Fig. 3*), is capable of switching even a ground terminal of RCA pin/jacks. So you can switch the ground terminal of inputs as well as the signal line to improve S/N ratio.

The 20 pins evenly aligned on the back makes it easy to install the connecting wires between the pins and RCA jack on the rear panel. I guess the only drawback is that all pins are placed on the same plate, so crosstalk between channels might be worse in the high-frequency range (above 100kHz) than with a split-plate rotary switch for each channel.

If you are concerned about this crosstalk, you might use a two-stage, 2-pole, 5-position rotary switch to reduce it, such as a unit made by Fujitsu, for example. But this CT-3 is very compact and well constructed, with a thick gold-plated electrode providing higher reliability than the regular product.

The CT-3 is furnished with a very attractive milled knob, made of stainless steel, which fits perfectly in my custom-designed case. Since I have never seen such a stainless-steel milled knob, such as you can find in a golf putter, this is really a nice feeling, because of its weight and surface finish.

## VOLUME CONTROL

The volume control (*Fig. 4*) also comes from DACT. It is the integrated product of the aforementioned rotary switch and a rounded copper-clad PCB with a number of surface mount resistor chips. The surface-mount resistor is used in a lot of handy and mobile equipment where small size is needed; for example, in mobile gear like Palm PC or Walkman, cellular telephones, and so on.

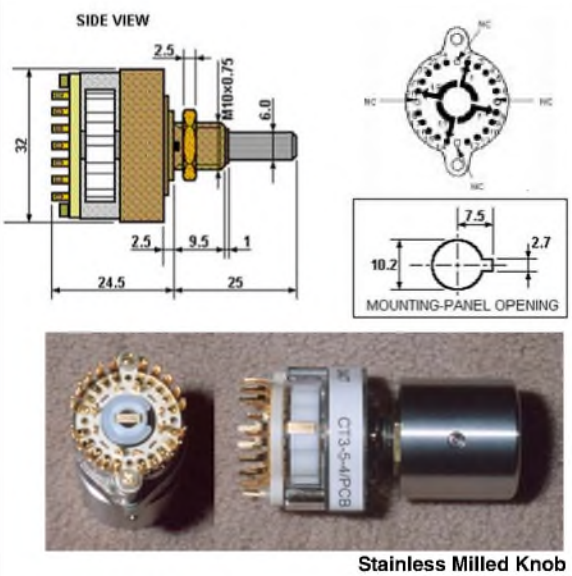


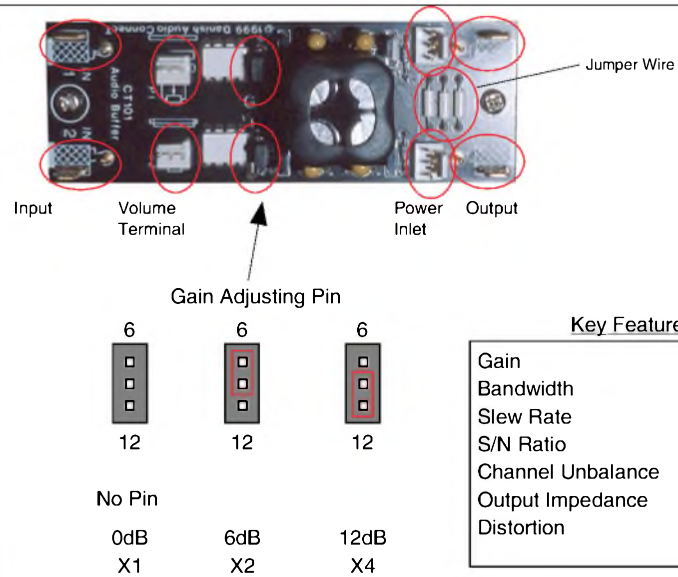

**FIGURE 3: CT3 drawing.**

The resistor itself is tiny to handle, but its size reflects the frequency response to be higher than ever—50MHz. The feeling when rotating the knob is also as good as the rotary switch. This is really better than the conventional switches that come with telecommunications equipment. I chose the value of 10kΩ to meet the output impedance of the EQ and buffer amplifier.

The volume itself provides a convenient feature that simplifies wiring between the volume and other components using a printed circuit connector as well as the buffer amplifier that mates up with this control.

The buffer amplifier CT-101 (*Fig. 5*) is very small, like a cellular telephone, with a frequency response up to 25MHz and flat in this frequency range. Also, three jumper pins on the board are available to select the gain (0dB, 6dB, or 12dB). This is a very convenient feature that I like.


**FIGURE 4: CT2 volume control.**




Key Features

| | |
|---|---|
| Gain | 0, 6, 12dB |
| Bandwidth | 25MHz |
| Slew Rate | 500V/S |
| S/N Ratio | 112dB |
| Channel Unbalance | ±0.05dB |
| Output Impedance | 0.1W |
| Distortion | 0.0002% |

**FIGURE 5: CT-101 buffer amplifier.**

Common features exist in both CT-100 and CT-101, since both modules use an operational amplifier:

1) They both need a complementary power supply, such as ±17–35V. Note that both modules provide a zener-diode-regulated ripple filter on the board, so that the internal voltage is stabilized at ±15V.

2) There is no need to solder hooking wires to the board, since printed-circuit connectors are used, and they come included with cables.

3) They feature low crosstalk through a power-supply line because of dual mono-amplifier structure.

4) Both feature ultra-low output impedance of 0.1Ω, bringing the high-hum immunity through an output line cable, and eliminating a shielded wire for the internal wiring. I did not use any shielded line cable inside.

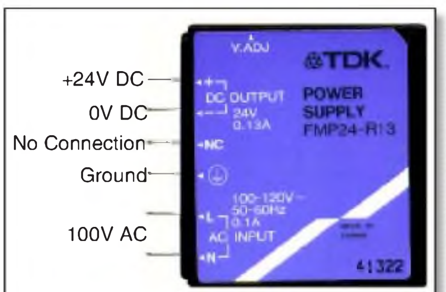



**FIGURE 6: Switching power module.**

## POWER-SUPPLY MODULE

The power-supply module (*Fig. 6*) is the first occasion to use a switching regulator as the ±24V DC power source. This is rarely seen in DIY types of amplifiers, whereas most digital circuits like PCs use this frequently because of its compactness and light weight. I happened to find this in Akihabara, suggesting that I could make this amplifier as small as a cookie case. The module—from TDK—is small enough to integrate into a small case.

The module produces no magnetic field; is small, compact, and light; and possesses a very wide range of AC line voltage between 90V and 120V, which allows this amplifier world-wide use compared to a conventional power supply with an E-I cored transformer.

The power consumption of the modules is 45mA and 5mA, respectively, so this power supply can drive both adequately because of its maximum supply current of 130mA. On the other hand, you might prefer a battery-powered supply if you do not like this switching power supply.

Two power modules are needed to drive these DACT modules because they require a complementary power supply. In addition, the AC line-voltage selection circuit offers 200 to 240V AC operation even in Europe. To do this, both power modules are connected in series with a couple of bleeder resistors to achieve the AC voltage even out of 200V AC inlet since the AC power consumption is less than 0.1A AC. Such a

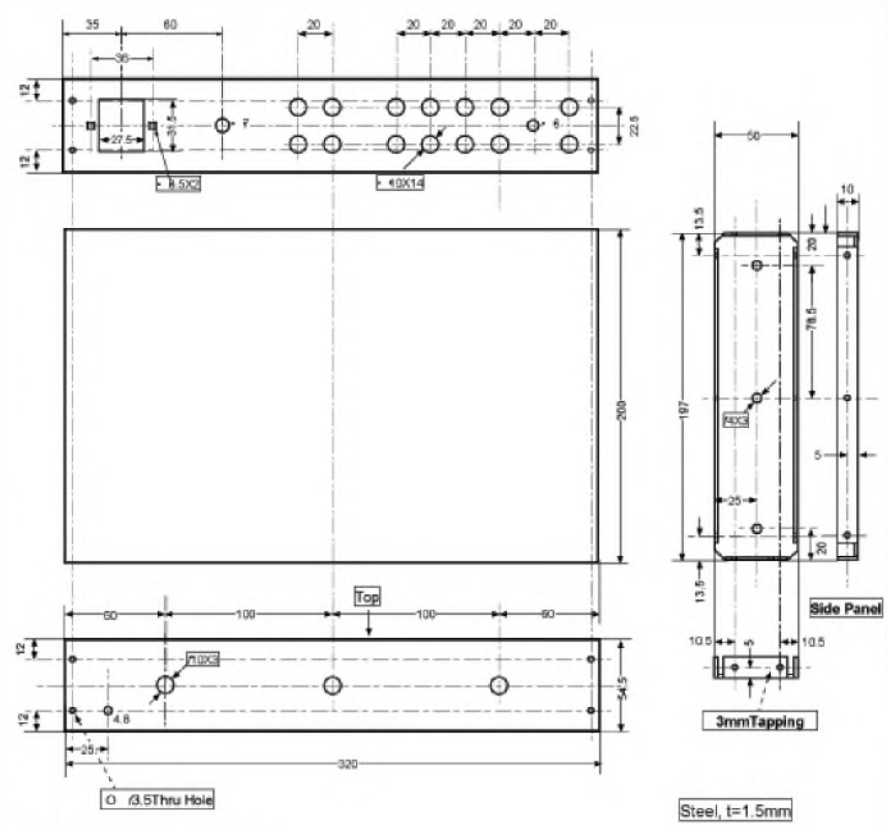



**FIGURE 7A: Case drawing.**

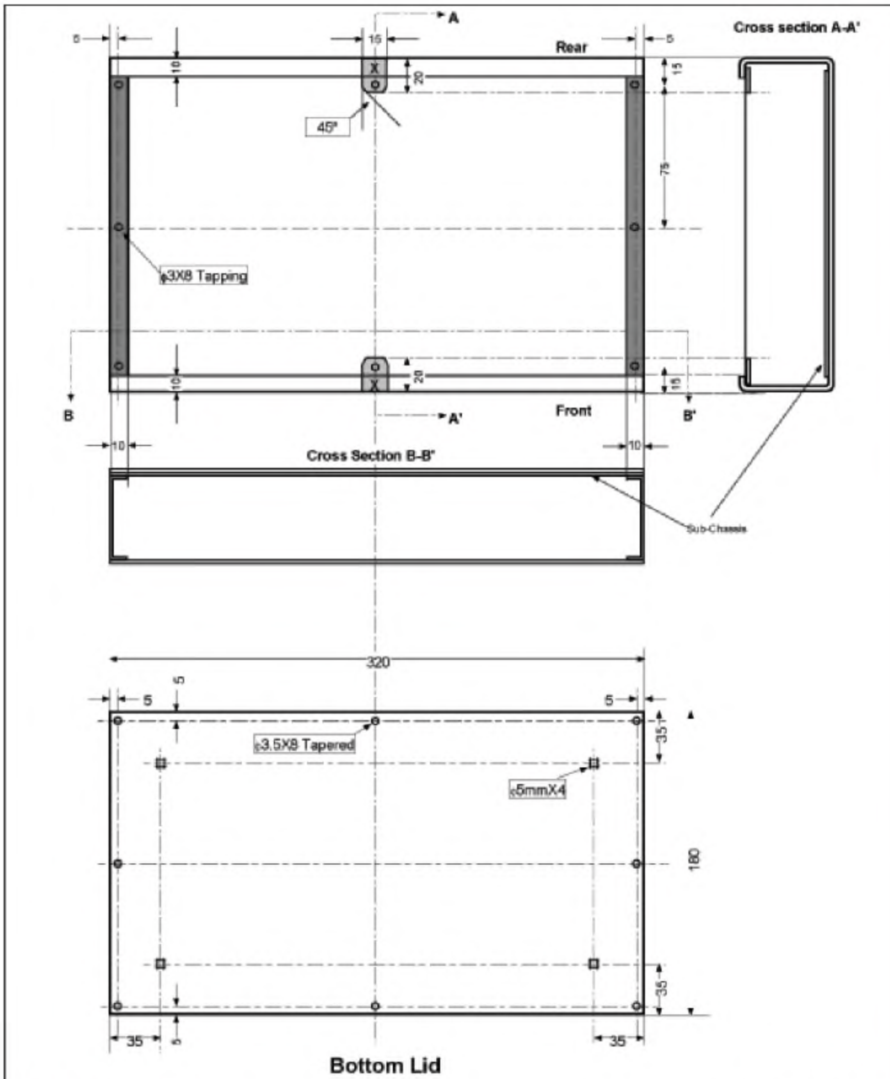



FIGURE 7B: Case drawing.

voltage delivery will probably work without any failure.

To make sure, I performed a 24-hour test under 200V AC operation as burn-in (regularly done at the semiconductor production stage), and I confirmed that no failure occurred under this configuration. This test result certifies that you can use this module anywhere in the world. If you use it only in Japan or in the US, where 100–120V AC is available, then you might dispense with the line-voltage selection circuit.

## ENCLOSURE DESIGN

For the case, I chose a size of 320 × 200 × 55mm (*Figs. 7a-7c*). The enclosure itself is shell-structured, with the top cover and the internal case using a 1.5mm thick stainless-steel plate. The major reason to use a top cover is to hide the screws that fix the parts over the inner case. The top cover of a stainless-steel plate conveys the impression of a manufactured amplifier and matches the stainless-steel knobs on the front panel. The drawback is making the case very heavy (2kg or so).

A wooden panel (200 × 58 × 9mm) of teakwood or similar quality is attached to the sidewall (*Fig. 8*). The panel was polished with #400–#600 sandpaper, drilled to fix to the metal side panel by three screws, and polished again with walnut oil. As a result, the surface of the wood became as fine and smooth as if it had been crafted by a professional.

I installed a rotary switch, a volume control (*Fig. 9*) and an AC power switch over the L-shaped subpanel separately built toward the back panel so the internal wiring would be shorter than usual. This structure requires an extension shaft, a shaft coupler, and so on, bringing the shaft of each component to the front panel.

## THE ASSEMBLY

First of all, disassemble the case, with

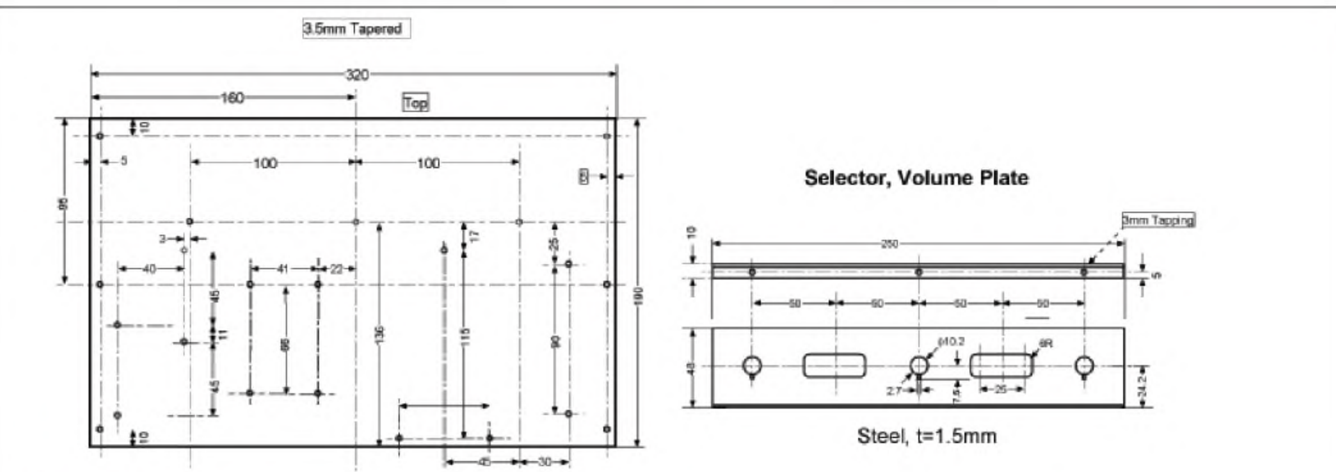



FIGURE 7C: Case drawing.

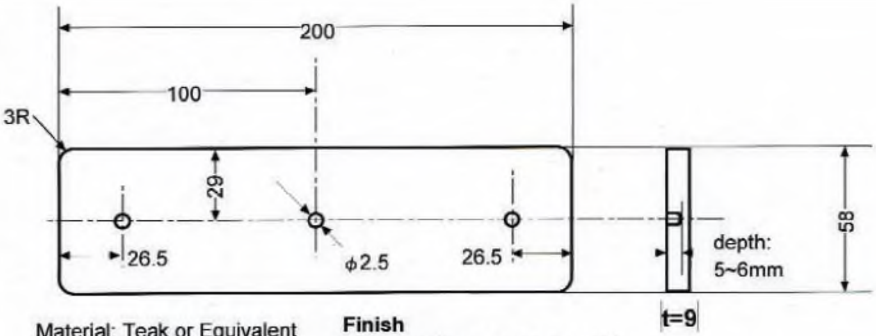



**FIGURE 8: Wooden side panel.**

7 pcs. of Walnut

its inner plate and side panels to be split. Second, affix power modules, an inner plate, and so on over the inner top plate. Third, attach a teakwood side panel over the side metal panel with three screws. Fourth, fix RCA pin jacks over the rear panel. Fifth, attach the volume control and switch onto the inner panel. Sixth, attach an extension sleeve, rod, and axes onto the front panel. Finally, fix the AC line inlet, pilot lamp and so on (*Photo 1*).

Soldering is needed only for the portion of the power supply and the input selector. To make assembly and maintenance as simple as possible, I devised a method of connectors and mating cables. The amplifier modules are linked with the volume control and power supply via such a cable (*Photos 2–3*, *Fig. 10*).

The input-selector section needs soldering at 48 points: the RCA input jacks and each terminal point of the rotary switch, using 3 to 4″ hook wires. This is the toughest job. To make this soldering easy, I strongly suggest fixing the hook wires to the terminal pins of the rotary switch before installing the switch in the internal case. The wire should be trimmed so its length reaches between the jack and its terminal (*Photo 4*).

Finally, connect each module using a hook wire with a connector. I believe even a beginner can manage this wiring and soldering, since the soldering can be done outside of the case (*Fig. 11*).

## ADJUSTMENT

No adjustment is required, but you should have a tester handy. You must check for wiring errors prior to turning on the power switch. Once you have checked and found no errors, disconnect the internal DC power connectors from the amplifier modules, then turn the AC power on.

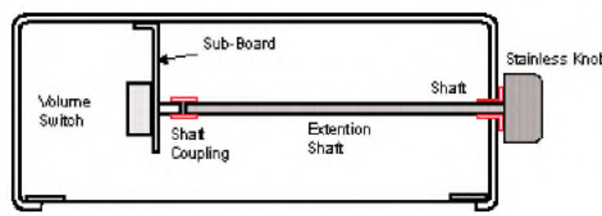



**FIGURE 9: Volume-control and switch installation.**

Check the power-module output-pin level, which should be +24V or so. The module output voltage was adjusted at the factory, though you can change this by adjusting the trimmer, which is placed on the side panel.

After this, turn off the power, plug the connector cable back in, and turn the power on again for standby (*Photo 5*).

## MEASUREMENTS

First of all, I used an inverse RIAA network to see how the measured RIAA curve compared with the theoretical curve. The network features the fully complementary frequency-response curve against RIAA equalizing circuit, and provides both 600Ω and 50Ω input terminals and –40dB and –60dB output terminals (*Figs. 12a, b*). The circuit is easily assembled with the selected capacitors and resistors (*Fig. 13*).

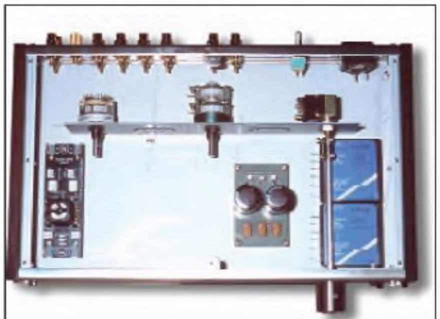


**PHOTO 1: Major components installed.**

The KF-1 circuit is easy to use since you can insert this network between an audio generator and the equalizer amplifier of CT-100, which then generates a flat signal after equalization. If the output signal shows a flat frequency response, it would be assurance that the amplifier works well (*Fig. 14*).

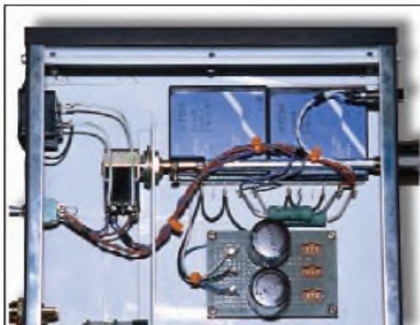


**PHOTO 2: Power supply.**

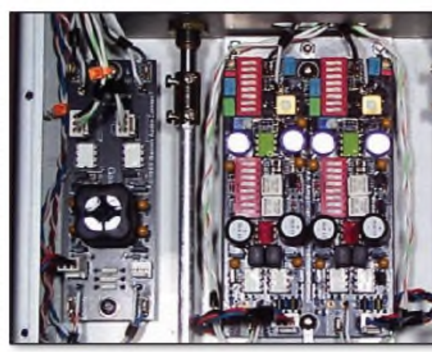
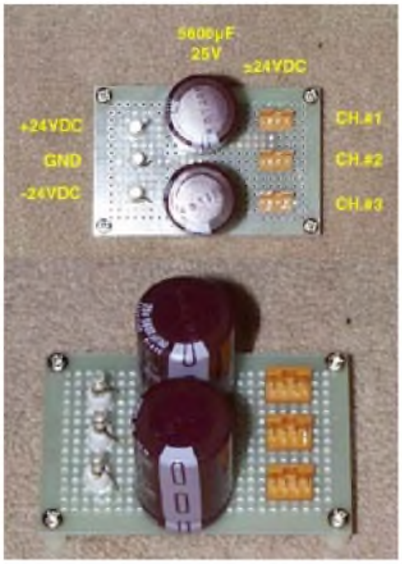


**PHOTO 3: Input selector.**



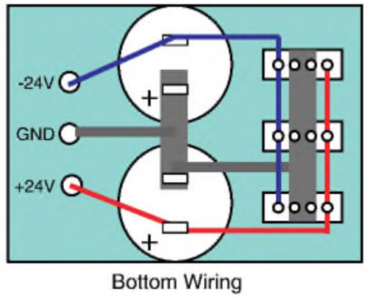




**FIGURE 10: DC power-supply board.**

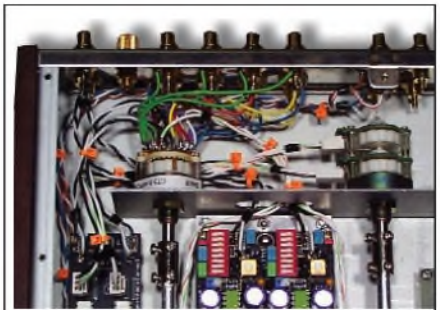


**PHOTO 4: Amplifier modules.**

- **Input versus output characteristics** (*Fig. 15*)
CT-100 showed very good linearity up to 10V output and also a stable gain factor of 6, for example, 600mV came out at the input level of 1mV. Also, CT-100 generates a maximum output voltage of up to 10V. CT-101 had as good linearity as CT-100, also showing the flat gain of 12dB (4× of gain factor) up to 10V.

- **Distortion** (*Fig. 15*)
CT-100 showed a worse distortion at 100Hz and 10kHz than at 1kHz by one digit, though the lowest value showed as low as 0.1%. It might be desirable that the optimized input sensitivity would have the output level between 1 and 5V. CT-101 showed a distortion below 0.1% in the 0.5–8V output-level range. I guess this is enough for regular usage, though at the output level over 10V, the distortion became dramatically worse.

- **Frequency responses** (*Fig. 15*)
CT-101 showed a very stable flat response over the frequency range from 10Hz through 1MHz and its deviation of gain is within 0.5dB. Even at a –6dB volume position against the maximum, the flat response curve showed performance to be stable.

- **Waveform** (*Fig. 16*)
Since CT-101 features a wide bandwidth of frequency-response curve of 25MHz, the square wave comes out without any distortion, overshoot, sags, and so on. On the other hand, the oscillator signal integrity might be a question. I do not see degradation of even 1MHz sine wave.

- **Residual noise voltage**
The measured residual output-voltage level was 0.1mV at both channels when the volume position was the minimum. Since it uses an operational amplifier, CT-101 generates a few mV DC offset voltage caused by a deviation in the fabrication of IC chips. I measured a 1.2mV DC offset level. So this might generate a pop noise through your speaker system when switching a selector. If this concerns you, I suggest inserting a capacitor into the output terminal to cut off this DC residual level.

Even when driving my homebrew 3CX300A1pp amplifier with this preamp, the residual output voltage at this main amplifier was only 0.5mV to 0.6mV, so that I cannot recognize any hum at all through my JBL S3100 speaker system.

Also, I have checked the residual noise of the CT-101 EQ amplifier by setting its input sensitivity to 0.25mV. The measurement of the output level of CT-100 through CT-101 was 0.2mV, 0.3mV at both channels when the volume position was the minimum. When tuning the volume level to the maximum, the output level at the main amplifier read several units of 10mV, though I could not hear any hum.

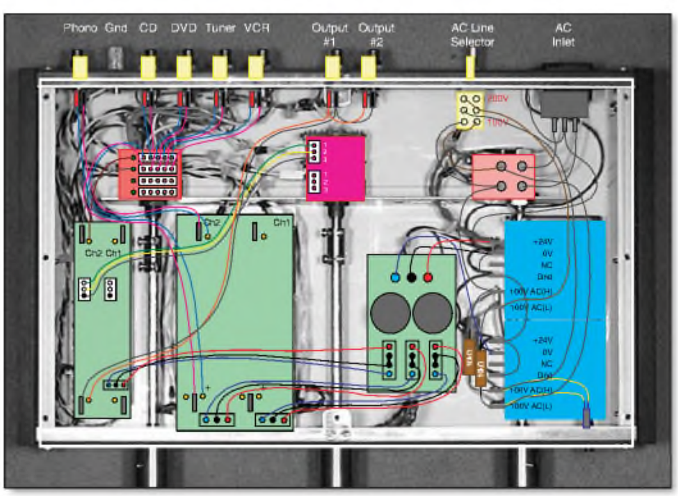



**FIGURE 11: Internal wiring.**

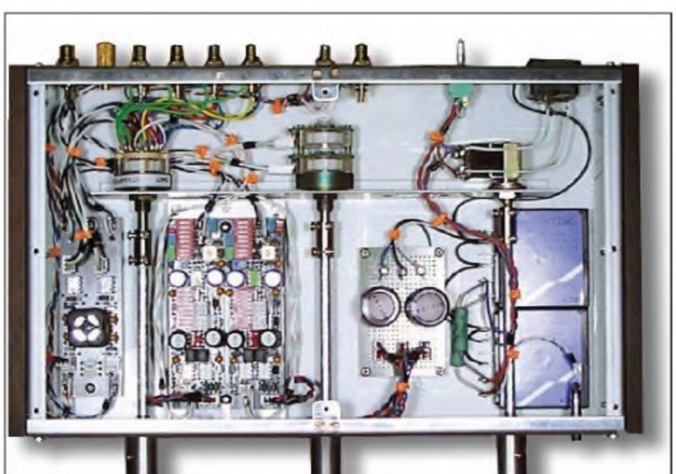


**PHOTO 5: Completed assembly.**

## LISTENING IMPRESSIONS

I brought this amplifier to a friend's home and tested with his JVC direct-drive turntable system and Ortofon MC-20 cartridge. First of all, the sound coming from this preamplifier was very impressive, since its sound is as clear and dynamic as if it surpassed the quality of a manufactured amplifier. The sound itself brought me no fatigue at all, even though I listened to all kinds of music for several hours, while a low-quality amplifier sometimes kicks my brain through my ears. The tone itself seems very soft and mild, though; it was dynamic, bringing more realism than if I were at the concert hall.

Changing the source from an LP to a CD, I checked the buffer-amplifier characteristics. This is really nice sound making me forget about the presence of

FIGURE 12A: KF-1 reverse RIAA filter.

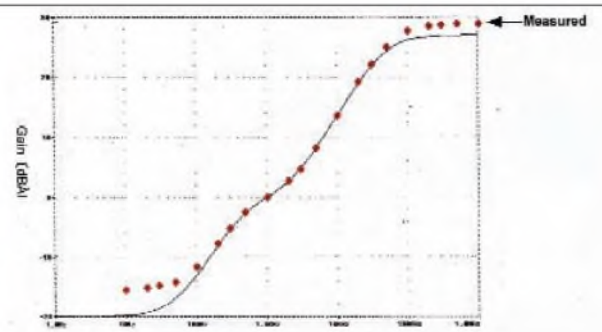



FIGURE 12B: KF-1 frequency response.

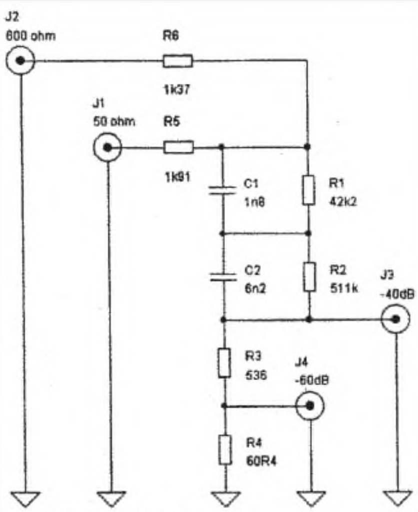



FIGURE 13: KF-1 circuit

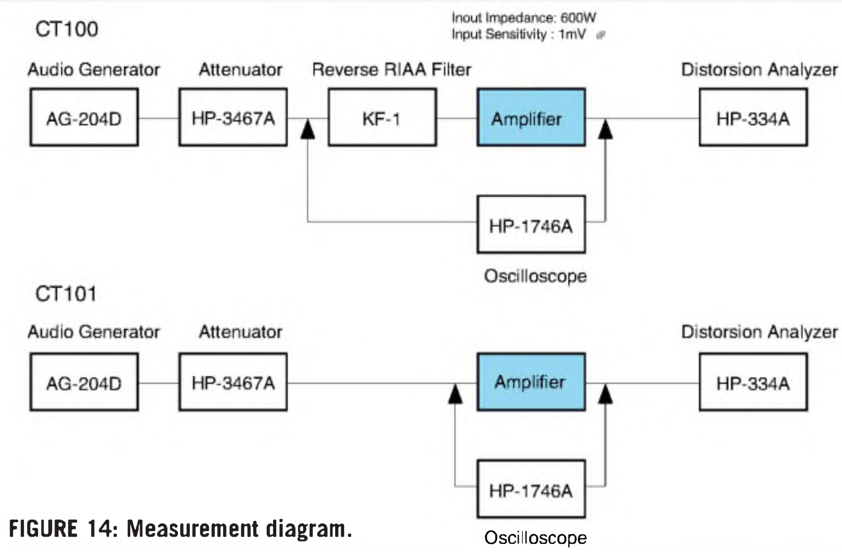



FIGURE 14: Measurement diagram.

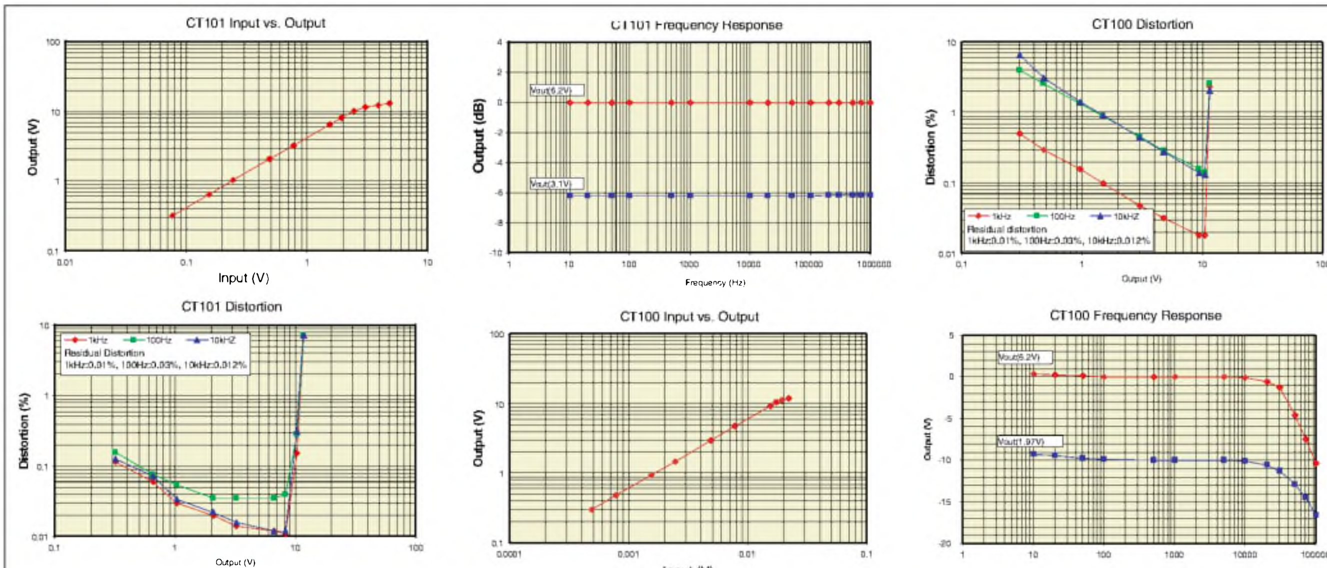



FIGURE 15: Characteristics.

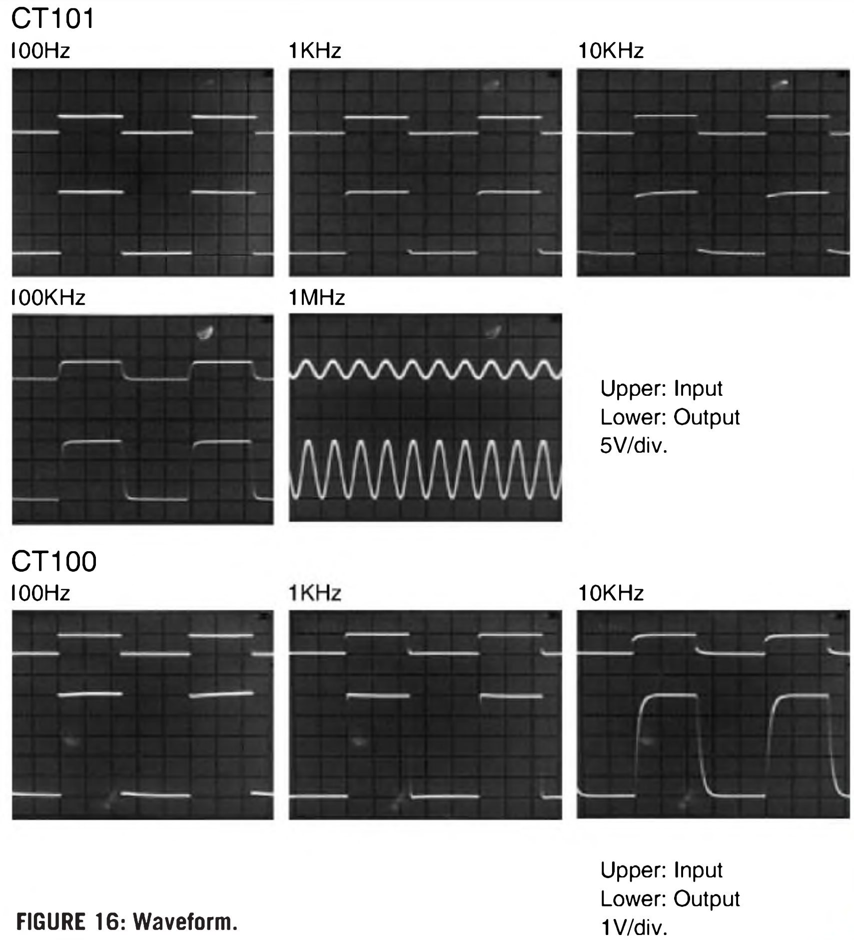



**FIGURE 16: Waveform.**

## TABLE 1
## PARTS LIST

| UNIT | PART | MODEL, SPECIFICATION, MANUFACTURER | PRICE |
|---|---|---|---|
| 1 | EQ amplifier module | CT-100, DACT | US $495[1] |
| 1 | Buffer amplifier module | CT-101, DACT | US$145[1] |
| 1 | Rotary switch | CT-3-5-4/wire, DACT, 4-pole, 5-position | US $62[1] |
| 1 | Volume control | CT2-10K-2, DACT, 10kΩ | US $136[1] |
| 3 | Knob | CT-knob1, DACT | US $43.50[1] |
| 3 | Extension Shaft kit | CT-ext1/S, DACT | US $46.20[1] |
| 1 | Custom made case | 320 × 200 × 55mm, stainless steel, San Ei Musen | ¥15,000 |
| 1 | Side wall | Teak, 200 × 58 × 9mm | ¥3,000 |
| 1 | AC power inlet module | 10A 250V, NICOON | ¥300 |
| 1 | Power switch | ARN01, 25A 125V, OTAX | ¥850 |
| 1 | AC line voltage select switch | 125V 6A, 2-pole, 2-position ML206, ALCO | ¥750 |
| 2 | Switching power module | 100V AC, 24V 0.13A, FMP24-R13, TDK | ¥800 |
| 2 | Electrolytic capacitor | 5600µF 25V, Nichicon | ¥600 |
| 1 | Epoxy PC board | SW-272, Sanhayato | ¥100 |
| 1 | PC board connector set | | ¥600 |
| 2 | Pilot lamp | Neon type | ¥140 |
| 7 | RCA pin jack | BS-015 | ¥4,900 |
| 1 | Metal feet | IAG (USA) | US $40[2] |
| 9 | Spacer, standoff | 3mm dia., 10mm long | ¥270 |
| | Screws & nuts | 3mm dia., hex nuts, washer, etc. | ¥500 |
| 1 | Hook wires | five colors 5m-roll | ¥1,500 |
| | | Grand total | ¥133,000 (¥80,000[3]) |
| | | w/o sales tax, $ = 107yen | |

(Notes)
1) Price at DACT direct, including the shipping fee from Bangkok
2) Price at IAG direct
3) Option: w/o CT-100, for the line amplifier

this buffer amplifier. Since this amplifier did not add any extra taste of sounds, it seems to me it is a monitor amplifier as well.

To close this listening test, when listening to the speaker system that I used, JBL S3100, I could no longer hear any microphonic noise. I think this is due to the extremely high S/N ratio of the buffer amplifier.

Furthermore, there was no hiss. Due to the ultra-low output impedance of 0.1Ω, the residual noise-floor level of the main amplifier driven by this preamp is extremely low, below 1mV. This shows the cleanliness of this amplifier, due to the premium DACT components.

I am very satisfied with the performance and the quality of this amplifier, so I am confident that anybody, even beginners, can build this amp and achieve the same quality and performance as a manufactured preamplifier—with simple assembly and less labor than ever before. ❖

### REFERENCES

CT-100, CT-101 Installation Instruction
DACT Technical Bulletin
Downloaded at http://www.DACT.com/

### MEASUREMENT EQUIPMENT

Audio analyzer HP-334A
Audio generator, Kenwood AG-204D
Reverse RIAA filter, Hagerman Technology KF-1
AC voltmeter, HP-403B
Attenuator, HP-3467A
Digital multimeter, Fluke 8020A
Oscilloscope, HP-1746A

### RESOURCES

**EQ Amplifier, Volume control, Rotary Switch**
Danish Audio Connect Thailand, Ltd.
15th Fl., Unit 1501/3
Ban Chang Glas Haus Bldg.
1 Sykhumvit Rd., Soi 25
Bangkok, 10110 Thailand
(+66 2) 260-6072
Fax: (+66 2) 260-6071
E-mail: info@DACT.com
URL: http://www.dact.com/

**Reverse RIAA Filter**
Old Colony Sound Lab
305 Union Street, PO Box 876
Peterborough, NH 03458-0876 USA
+1-603-924-6371
Fax: +1-603-924-9467
E-mail: custserv@audioXpress.com
URL: http://www.audioXpress.com/

**Others (Tokyo, Japan)**
Suzusho
Radio Store
Neji-No-Mizutani
Taiger Musen
Tokyu-Hands
Abo-Denki

# The DR12 Horn

This author presents another breakthrough in speaker design: a 12″ folded-horn without high-frequency rolloff.

**By Bill Fitzmaurice**

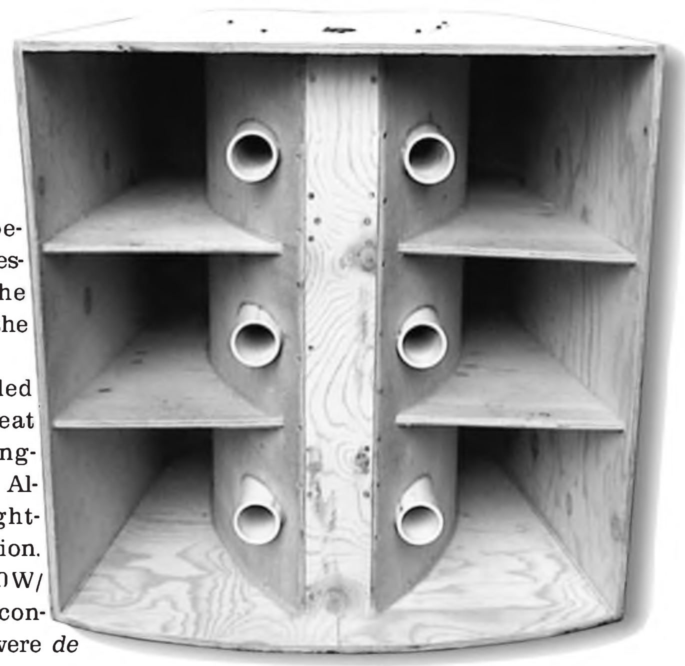


**PHOTO 1: The completed DR12.**

In the movie *Back to the Future*, there is a scene where Marty McFly straps on his axe and fires up his amp. The camera pulls back to reveal a circular object some 8′ in diameter against the wall. It is a speaker. Marty cranks up the volume on his axe. A 60-cycle background hum begins to shake the room. He strums a chord. The resulting sonic quake picks Marty up and throws him against a wall as a pile of rubble buries him.

Hollywood fantasy? Maybe, but if you were to crank up your axe to the max through one of my DR12 cabs, it would be loud—very, very loud.

## SUPERIOR PERFORMANCE

Another in my line of prototypes for a manufacturer, the DR12 evolved from the DR10 (January '01 *aX*), which gives a performance superior to any other single-ten box. In designing a 12″ driver version, I could merely have increased the dimensions as required and achieved a good result. But the target market for the "12" cabinet was concert-touring sound, with an anticipated price of at least $1,500. For that kind of money, the new design would need to be clearly superior in both sensitivity and bandwidth, to the direct radiators that currently dominate that market segment.

There are about a dozen companies that have a stranglehold on this lucrative market, and the only way a newcomer could get any business would be through building the better mousetrap. It promised to be a challenge indeed. Yet, in view of how well the DR10 worked, I was certain I was on the verge of another breakthrough in folded-horn design, hopefully one that would resurrect the horn as the cabinet of choice for the pros.

The last horn-loaded cabinet to enjoy great favor in the touring-sound genre was the Altec™ A-7, a straight-horn/reflex combination. Back when 100W/channel amps were considered large, these were *de rigueur* for large-venue work. Due to an oversized pre-T/S reflex section, they were inefficient from a size standpoint, while their labor-intensive construction made them expensive, ultimately dooming them for touring. Though you'll still find more than a few horn cabinets hiding behind the screens in movie theaters, what most manufacturers produce today for touring sound are simple T/S boxes.

The proliferation of high-wattage (over 1,000W/channel) amps capable of running into 2Ω loads (or less) has made the job of designing high-powered cabinetry a fairly simple affair: stuff a few beefed-up woofers into inexpensive ported boxes, using horns for mids and highs only. Manufacturers content themselves with 100dB/2.83V SPL ratings, since combining lots of cabinets with powerful amps will get the required output. Three-way systems are the norm, and four-ways are not unusual.

When using bi-, tri-, or even quad-amping, it's not too difficult to get 140dB output, provided you have enough cabinets, the amps to drive them, and roadies to carry them. Touring sound companies routinely carry 20,000W of power in their racks to serve audiences of 5,000 or less, using 18-wheelers and 20-man crews to lug the cabinetry. (Contrast this with Woodstock in '69, where an audience of at least 250,000 was served by only 10,000W, thanks to horn-loaded cabinets.) Lots of cabs, lots of amps, crews of roadies, and fleets of semis add up to a lot of overhead. Now you know why concert tickets costing less than $50 are about as common as nickel cokes.

There are full-range horn systems still to be found, notably in the UK. Prompted by $4-per-gallon gasoline and trade-union wages for roadies, horn-loaded cabs from companies such as Turbosound™ have reduced both the wattage and tonnage requirements for concert sound. However, their limited bandwidth designs still require using complicated three-way triamped or four-way quadamped configurations.

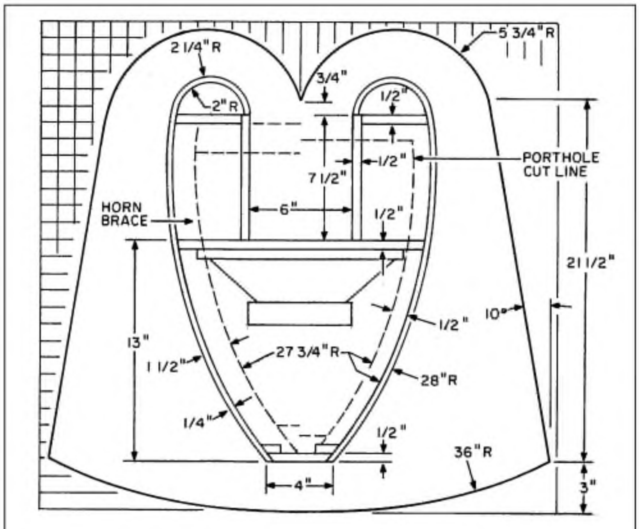


**FIGURE 1: Top view.**

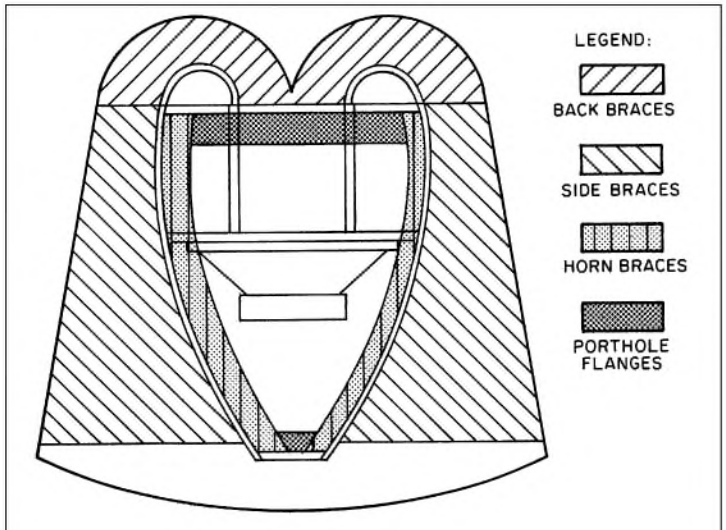


**FIGURE 2: Braces.**

## SMALLER, LOUDER, BETTER

The motto for my designs is "smaller, louder, better." I also like the idea of simpler. I wished to devise a speaker system simple enough to provide even budget-conscious garage bands having limited technical skills with sound as good, and as loud, as the best contractors can deliver, but without complicated crossovers and multikilowatt amps. To that end, my DR12-design requirement was to produce a two-way cabinet of less than 15ft$^3$, capable of 130dB output, 100Hz to 16kHz, from 300W input, without biamping (*Photo 1*).

Getting the necessary flat response and bandwidth from a woofer to accomplish this task required a re-examination of why folded horns have never before equaled direct radiators in those respects. A cross section of any folded-horn cabinet appears as a quasi-tuba, with the driver serving as the mouthpiece. Because of the limitations of materials and construction methods, traditional folded horns consist of a number of flat panels, whereas a tuba, or any brass instrument, has nothing but smooth curves.

Four years ago I postulated that the restricted high-frequency response of folded horns was due to the use of flat panels and angled corners that caused out-of-phase reflections within the horn. When I eliminated as many angled corners and flat panels as possible in my Snail series of folded horns, replacing them with curved surfaces, high-frequency performance was greatly improved. The DR10, which evolved from my Siamese Snail, has a bandwidth fully two octaves higher than was previously thought possible.

When designing the DR12, I posed a question to myself: If eliminating angled corners and minimizing flat panels gained two additional octaves of bandwidth, how could I gain even more? Since it was evident that the greatest potential for phase problems was in the bend of the horn, I turned my attention to that area and devised a horn bend that consists of offset concentric arcs, allowing true horn shape. When viewed from the top (*Figs. 1* and *2*) the cabinet interior really resembles a pair of tubas back-to-back. The sonic result of this configuration is the virtual elimination of out-of-phase cancellations within the horn bend.

## FLARE RATES

Like the DR10, the DR12 consists of two horns in series, connected at the bend. Whereas traditional horns have a single flare, DRs have two flare rates. To load highs, the flare is rapid near the driver cone, in what I call the "throat horn." This section could serve as a stand-alone midrange horn, and, not coincidentally, that is what it looks like. Once past this initial stage, the horn splits into dual "mouth horns," where the flare slows to load the lows. This dual-flare arrangement loads the driver over a much wider bandwidth than a single-flare horn.

DRs have larger throat openings than traditional horns, differing from Snails, where I used small throat areas in an attempt to better load the highs. While not abnormally small according to prevailing theory, those throats caused driver Fs(h) to go too low. For example, take the Snail II, loaded with a JBL-E140™ driver with an Fs of 32Hz. A 45 in$^2$ throat resulted in an Fs(h) of 20Hz, the consequences of which can be seen in *Fig. 3*.

Note how the Snail II has better response than the DR12 at 32Hz, but a severe response dip at 62Hz. This is caused by needing to tune the vented section of the cabinet well over two octaves below the passband of the horn in order to match the very low Fs(h). The DR12, with an Fs(h) of 32Hz, allows tuning the reflex section only 1½ octaves below the horn passband, eliminating the dip.

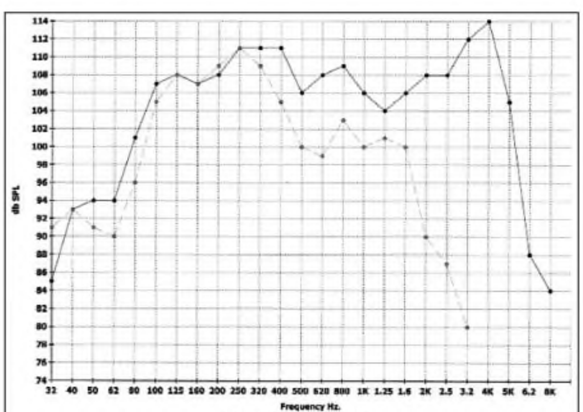


**FIGURE 3: Snail/DR12 response comparison.**

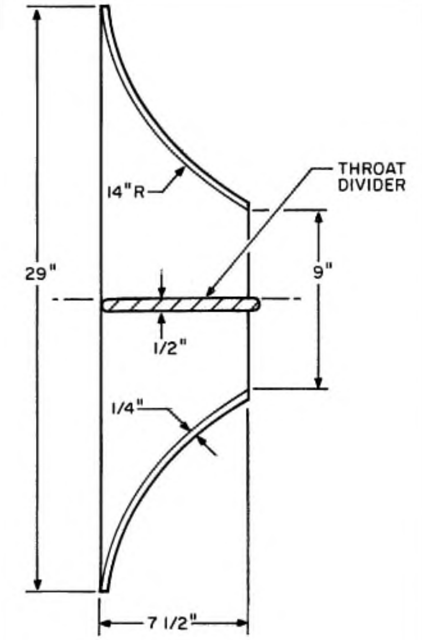
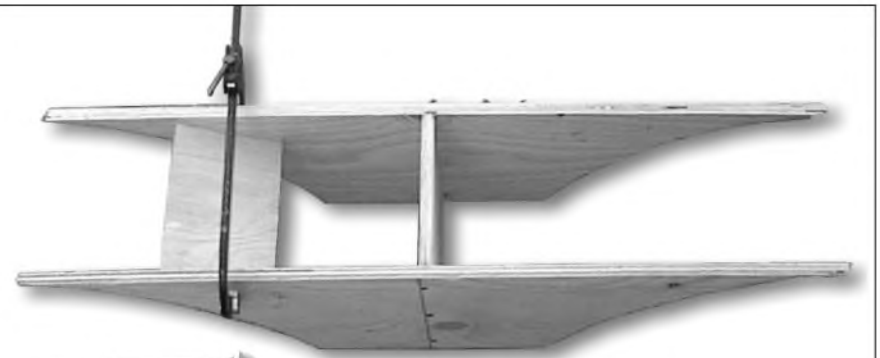



**FIGURE 4: Throat horn—side view.**



▲ **PHOTO 2: Clamping the throat horn.**

◀ **PHOTO 3: Applying the sheathing.**

## DESIGN

As in the DR10, the DR12 vents its rear chamber via ducts that exit into the horn mouth. I wasn't particularly enamored with the look of the ducts on the DR10, so I altered the mounting configuration for sleeker lines. (If you've built or plan to build any DR10s, they can easily be altered to the new scheme.)

Also like the DR10, the DR12 box is trapezoidal, for setting multiple units in a seamless semicircular array. Though inefficient space-wise, I included the trapezoidal shape as a concession to marketplace realities. Most professional sound contractors set up multiple cabinets in semicircular arrays in an attempt to broaden horizontal coverage. Manufacturers have encouraged this practice by making cabinets with trapezoidal shapes, and you can hardly find a high-end PA cab without this feature.

While it seems logical to set up cabinets in this way, a series of response peaks and nulls are produced when multiple high-frequency sources are placed on a horizontal plane. Maximum high-frequency horizontal dispersion is achieved by a vertical tweeter array. For that reason, tweeters in the DR-series cabinets are vertically arrayed, and you should also stack vertically multiple cabinets as far as possible.

Since the DR12 is intended as a large-venue concert-grade speaker, it is able to house a premium 12″ driver. I used an EVM-12L™, with nominal specs of Fs 55, $Q_{TS}$ .25, $V_{AS}$ 2.5ft$^3$, and SPL 100dB. Other premium drivers, such as the JBL-E120™, will work, as will midline MI drivers, as long as the specs are reasonably close. The main recommendation for the EVM™ is its superior high end.

The most important driver spec is the Fs. Do not use a driver with an Fs of less than 45Hz or more than 60Hz. The well broken-in EVM I used is over 20 years old, has been in more cabinets than I can remember, and consequently has an Fs of 48Hz. When mounted to the throat horn alone, the Fs(h) dropped to 40Hz; mounted in the completed cabinet, the Fs(h) is 32Hz. The preferred Fs(h) range for this cabinet is 30 to 45Hz. A driver Fs of more than 60Hz will kill response in the first (32 to 64Hz) octave, while an Fs of less than 45Hz will cause a response dip such as in the Snail II.

## CONSTRUCTION

As in all my cabinets, the prototype was built mainly from ½″ plywood. Thicker material is just not necessary with this self-bracing design. However, you may opt to use ⅝″ plywood for its easier joinery aspects, especially where the horn braces are concerned. Don't use ¾″ plywood, however. While the prototype weighs in at a manageable 80 lbs, ¾″ plywood would increase the weight to over 100 lbs with no sonic improvement.

You may fashion the sheathing for the throat horn from ¼″ plywood if it has been kerfed every 1″ or so. However, the tight radius on the cabinet back requires using two layers of ⅛″ plywood laminated together there. Since you must buy a sheet of ⅛″ ply

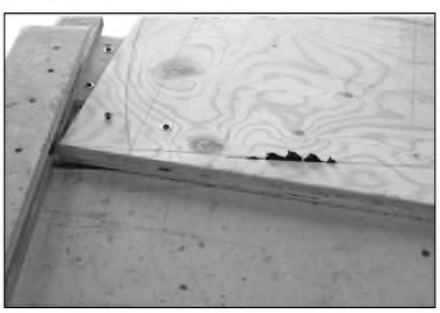
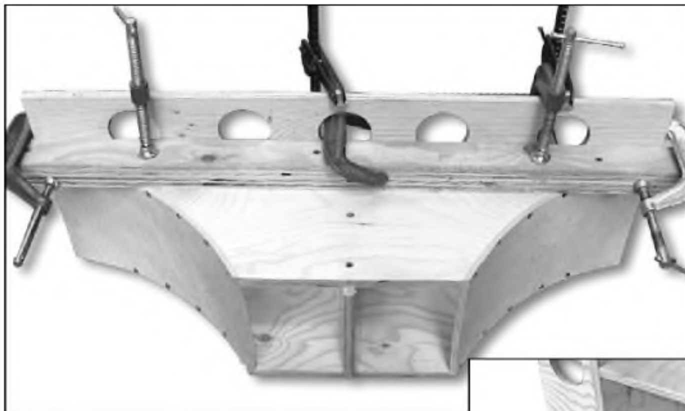


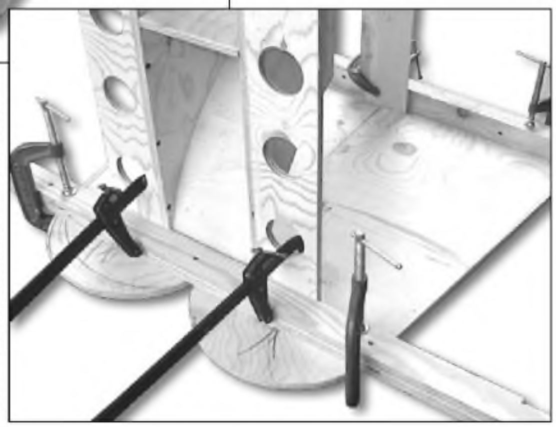
**PHOTO 4: 10° off-center cut.**

**PHOTO 5: Cutting throat overhang.**



◀ **PHOTO 6: Attaching horn supports.**

▼ **PHOTO 7: Attaching throat horn to top.**



for the back, you might as well use it for the throat horn as well. The mouth horn sheathing uses ¼″ plywood, since its bending radius poses no problems, though laminated sheets of ⅛″ will work and would be stiffer.

Secure all joints with drywall screws, using 1¼″ except as noted. Pilot-drill and deeply counter-bore all screw-holes. Glue and seal all joints with construction adhesive, for airtight joints are critical. I have recently started using a polyurethane construction adhesive, and find that it works better than older types, especially when it comes to sanding off, because it cures harder. You'll also need a hot-melt glue gun for some areas where quick setting is required and strength is not problematic.

All horn arcs have true radii. To draw them, make a "compass" from a long plywood stick. A drywall screw passes through one end of the stick to serve as a pivot, while ¼″ holes are drilled through it at the required radius distances. Insert a pencil through the appropriate hole to draw the arcs.

## BUILDING SEQUENCE

Construction begins with the throat horn (*Fig. 4*). Cut the sides and divider from ½″ plywood. Here, and in any case where two identical parts are required, it's best to rough-cut two pieces a bit oversize, screw them together, and finish-cut them both at the same time. The divider extends ½″ beyond the sides towards the baffle, and both ends of it are rounded either by sanding or routing with a rounding bit.

To minimize phase problems, the two halves of the throat horn should be not quite identical, so offset the divider slightly to one side, attaching it with one edge aligned along the centerline of first one horn side, and then the other. Clamp a piece of plywood scrap 4″ wide into the assembly, to hold the parts in alignment when the sheathing is applied (*Photo 2*).

Cut a total of four pieces of ⅛″ plywood for the sheathing. Liberally coat one piece with adhesive and layer it with another piece, then clamp the two pieces to the assembly and glue and screw them into place. There should be about ¼″ of overhang at each end. Repeat the procedure on the other curve (*Photo 3*). Put the assembly aside while the adhesive sets.

Lay out the top and bottom, drawing on both of them the location of all intersecting parts. Rough-cut the two pieces, screw them together, and then finish-cut, except for the arcs of the back. Cut the side edges at a 10° angle with respect to the centerline. You can do this easily by first cutting 10° plywood wedges on the table saw, using the miter gauge, and screwing them to your panel-cutting jig. (If you haven't made one yet, do so right now. Cabinets of this level of sophistication cannot be accurately made with freehand cuts.)

Put the piece on the jig, slide it through the saw, and you have a perfectly straight cut 10° off center (*Photo 4*). When you're ready to cut the back curves, screw a piece of plywood for the back brace to the unit and cut all three parts at once, marking each piece as to left and right side of the cabinet. This will make aligning parts a lot easier later on. After separating the parts, mark the cut line for the porthole on the cabinet top and cut it out.

Once the adhesive on the throat horn has set, you can run it through the saw (again on the panel-cutting jig) to remove the excess on the ends of the sheathing (*Photo 5*). Use a belt sander to trim the excess at the throat opening. Cut the horn supports, and drill randomly spaced 2″ holes through them with a hole saw. Sound waves hate flat surfaces; these holes reduce the flat-surface area within the cabinet.

Attach the horn supports to the assembly. Here (and on all joints) first clamp a straight guideboard to the joint line and clamp the mating part to it. Once everything is aligned perfectly, glue and screw the joint (*Photo 6*). Clearance here is a bit tight, and screwing will go much easier if you use a small right-angle ratcheting screwdriver.

## THROAT HORN AND TWEETER BAFFLE

Attach the throat horn to the top (*Photo 7*). Cut out the tweeter baffle and attach it also to the top. The sides of the tweeter baffle are cut at a 35° angle; cut this from a piece of plywood about 7″ wide, leaving two slivers about 1½″ wide each, with a 35° cut on one side as selvage. Save these scraps for later use.

Attach the bottom to the assembly. Cut the baffle to size (with each edge cut at an 8° angle), hold it in place in the cabinet, and through the throat horn trace the location of the throat hole. Note that the baffle attaches to the cabinet bottom, but not the top, extending past the top of the throat horn only as far as necessary to accommodate the driver. Cut the throat hole, and also drill some random holes as was done to the supports.

To the baffle, attach a driver spacer fashioned from ⅛″ or ¼″ plywood to prevent cone slap, and drill it and install T-

PHOTO 8: Attaching baffle to assembly.

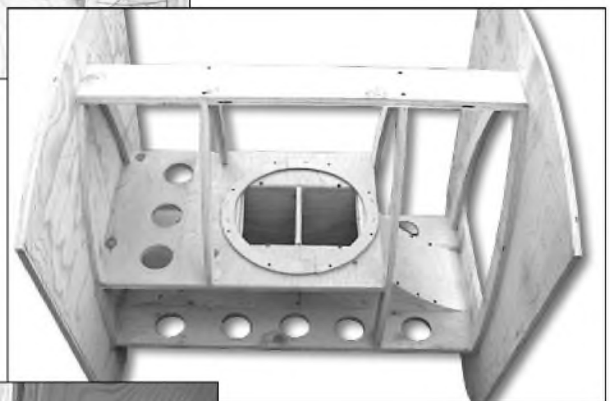
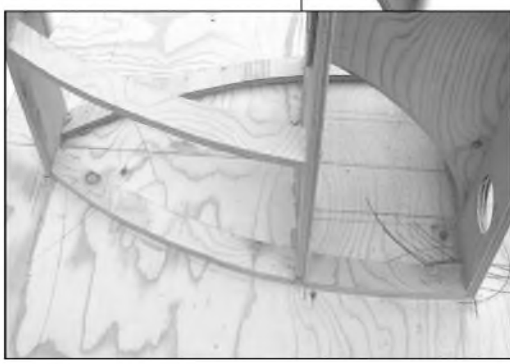


PHOTO 9: Braces.

PHOTO 10: Closeup of bracing.



nuts. Attach the baffle to the assembly, checking with a carpenter's square for proper alignment (*Photo 8*). Drive one screw on an angle through the horn sheathing to the baffle, and use plenty of adhesive for an airtight seal.

Cut one horn brace, comparing it against the assembly at the cabinet top for proper fit, and then duplicate it seven times, for a total of eight braces. The braces that attach to the cabinet top remain whole, as does the set immediately below them that intersects the top edge of the baffle. The remaining two sets must be cut where they intersect the baffle. You may omit the two smaller sections that would span from the baffle to the supports (*Photos 9* and *10*).

When attaching the braces that extend between the baffle and the tweeter baffle, have your driver in place so that you are sure to avoid locating the braces where they might interfere with either the driver frame or access to the driver mounting bolts. At this time also trim the braces as necessary to be sure you can install and remove the driver. (If you are using a JBL-E120, you may remove the rubber magnet cover to increase clearance inside the cabinet.)

Cut the 35° edged scraps to size to fit both sides of the rear of the tweeter baffle between the braces, and install them, making the tweeter baffle a full inch thick on either edge. Cut the mouth-horn sheaths and install them, attaching them first to the tweeter baffle, pulling them into shape against the supports with a long clamp (*Photo 11*). When the adhesive has set, sand the mouth-horn and throat-horn sheaths flush to the horn supports, and the mouth sheaths with the tweeter baffle, rounding this joint a bit.

## REFLECTORS

The inner reflectors are made of 4″ Schedule 40 PVC pipe that has been cut in half lengthwise (*Photo 12*). Place the back braces against the top (or bottom) and use a short piece of halved PVC to mark on them where to cut two semicircles that will allow the braces to fit over the reflectors. Using 3″ drywall screws, temporarily screw the reflectors to the horn supports. Place the back braces over the reflectors, spacing them not quite evenly between the top and bottom, again so that reflections between these parallel parts will not occur at the same frequency.

Place a straightedge (or straight piece of wood) across the back from the top to the bottom to correctly align the back braces (*Photo 13*). Use hot-melt glue to firmly secure the braces to the reflectors only. When the glue is thoroughly set (after about ten minutes) remove the reflectors and put a couple of reinforcing 1″ drywall screws into each back brace from inside the PVC.

Loosely fill the PVC with poly-fill, and then reinstall the reflector/brace assembly, this time using hot-melt glue to attach the reflectors to the supports and the back brace to the throat-horn divider; the rounded edge of the reflector will serve as a trough for glue. Go over the joints between the reflectors and the supports with sufficient hot-melt to be absolutely sure of an airtight fit.

The ducts may be cut from either 1½″ or 2″ PVC pipe. The prototype used 2″ to prevent "chuffing" at very high

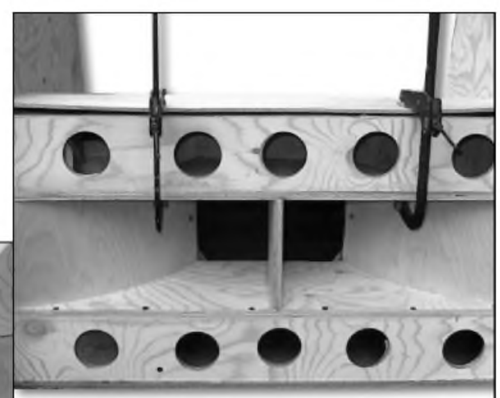


PHOTO 11: Installing mouth-horn sheathing.

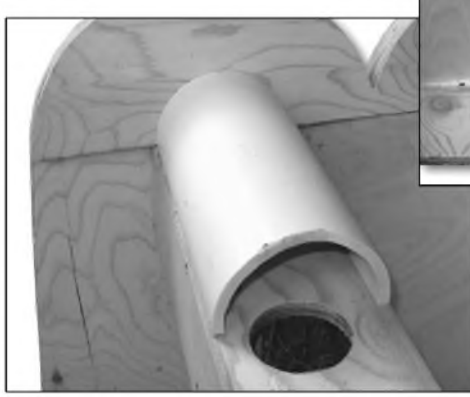


PHOTO 12: Applying PVC pipe.

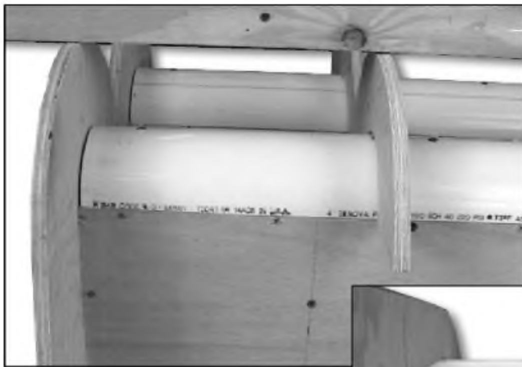


PHOTO 13: Aligning back braces.

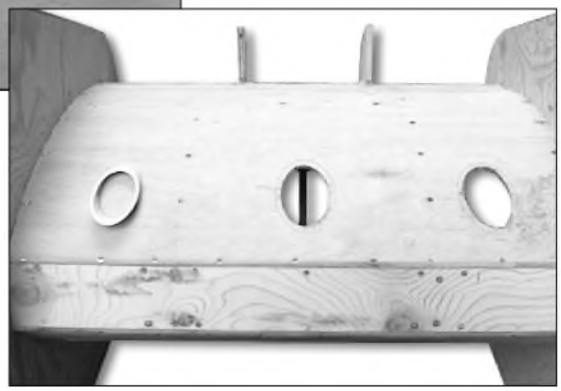


PHOTO 14: Duct holes.

levels, but 1½″ ducts will do in a pinch, and won't need to be as long. Cut a sliver of pipe at a 30° angle, using it to trace onto the mouth-horn sheaths the cutting holes for the ducts, which are evenly spaced top to bottom (*Photo 14*). After cutting the holes with a fine-tooth saber-saw blade, use a 2″ drill-mounted sanding drum to chamfer the holes for a good fit with the ducts.

Rough-cut the side braces about ½″ oversize. Temporarily clamp them in place, butting the back braces, and trace on them the true contour of their joint with the sheathing; if the gap is very wide, hot-glue your pencil to a sliver of plywood as a spacer (*Photo 15*). Cut the curves and clamp them in place again, this time running a straightedge between the top and bottom to mark the exact location of their outer edges. Finally trim the braces and install them, driving two or three screws into them from the inside of the cabinet (aided again by using a right-angle ratchet driver).

## BACK SHEATHING

The back-sheathing halves are installed much like the throat horn sheathing, with two pieces of ⅛″ plywood installed simultaneously, generously slathered with adhesive between them, using 1″ drywall screws every 3″ or so. The back sheathing should extend at least 2″ past the curvature of the back. You cannot use straight clamps to pull the back sheathing into place for screwing, although a strap-style wrapping clamp would work well. In any event, ⅛″ plywood flexes easily enough so that you can bend it and hold it with one hand, while drilling and screwing with the other.

Clamp together the two layers of the exposed ends of the back pieces, using scrap plywood cauls to aid in the process (*Photo 16*), until the adhesive sets. Use plenty of adhesive to fill the joint between the two halves of the back. When the adhesive is set, remove the clamps and cauls and use a circular saw, with a shallow-cut depth, to trim the ends of the sheathing so that both layers are flush.

Cut the sides so that they will extend to the curve of the back, overlapping the sheathing (*Fig. 5*). Use a router to produce a ¼″ rabbet on the sides where they overlap the sheaths. Round off the rear edges of

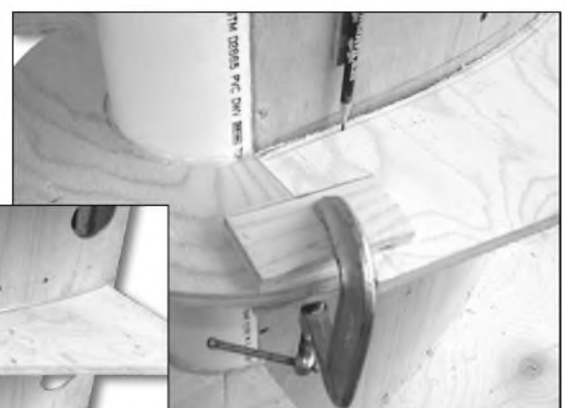


PHOTO 15: Side braces.

PHOTO 16: Applying the back sheathing.

## MEASURING HORNS

Fs(h) is the resonance point of the driver/horn combination with the rear of the driver open to free air, which is lower than the driver Fs. This fact has not been sufficiently addressed in folded-horn designs. To measure Fs(h), the driver rear must be unenclosed, an impossibility in most conventional folded horns, where intrinsic to the horn is a panel that seals the driver chamber.

Since the design of folded horns made Fs(h) measurement impossible, it was a factor that designers, including myself, did not consider. This measurement became possible when, starting with the Snail II, I started using porthole driver access, allowing the driver to see free air on the cone rear while the horn remained intact. Since this measurement did not appear in previous literature, I did not consider it until experimenting for the DR10 project.

For the flattest response, Fs(h) should be about one octave below the Fh (flare frequency) of the horn. Flat response to 32Hz thus would require an Fh of 64Hz, and consequently a very large cabinet. A spread of 1½ octaves is adequate for live-performance speakers, which need not have flat response below the horn passband. Close to a two-octave spread can still give an acceptable result.—**BF**

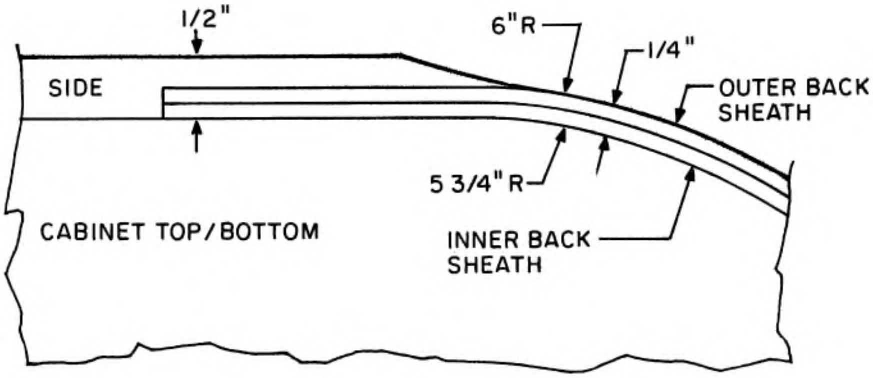



**FIGURE 5: Side/back joint detail.**

**PHOTO 18: Jack location.**

the sides for a smooth transition between the parts, and install the sides (*Photo 17*).

Cut holes for the tweeters in the tweeter baffle. (The prototype was built without tweeters and left with a natural finish, since it will likely never actually see service.) I would suggest using two Motorola™ Twin-Bullet KSN-1177 piezos, which total four elements for very high sensitivity. If you choose other piezos, use three elements for flatter response, up to six for long-throw applications. Mount the tweeters as close as possible to each other on the baffle, being careful that they do not hit the woofer magnet.

Cut scrap plywood to complete the flange around the porthole, weather-stripping it. If you prefer to use bolts and T-nuts to secure the porthole, install the T-nuts now. Install the woofer, wiring it to your choice of jack. The woofer fit is not severely tight, but you must reach through the duct or tweeter holes to bolt it in place. The porthole is a good spot for the jack (*Photo 18*) if you plan to use one cabinet per stage side. If you wish to stack cabinets, place the jack on a cabinet side. You'll need to drill a hole through the horn sheath to feed the wire through, being sure to caulk the hole airtight.

**PHOTO 17: Installing the sides.**

## TESTING

Now is the time to test for the box Fs(h). Hook up the cabinet to a function generator. With the porthole open, sweep-test the impedance for a peak in the vicinity of 40Hz. This Fs(h) will also be the box frequency, Fb. In *Fig. 6* (dotted trace) note how the prototype has a strong Fs(h) peak at 32Hz. With the Fb tuned to 32Hz (solid trace), the peak is replaced by a trough, flanked by two smaller peaks above and below it. (The next impedance low, at 100Hz, is the Fh, the beginning of the horn passband, above which both impedance and sensitivity rise rapidly. The peak at 125Hz is a one-fourth wavelength resonance of the horn mouth. At 160Hz, the impedance settles in at an average of 9Ω, until rising voice-coil inductance becomes apparent at 1.2kHz.)

Install the tweeters, weather-stripping the frames, wiring them parallel in-phase to the woofer. Loosely but thoroughly stuff the cabinet with polyfill, making sure to fill all voids, and install the porthole cover. Start installing the ducts. The middle ducts cannot be longer than 5″ or so, or they will hit the driver magnet, while the lower ducts are limited to about 11″ before they will be too close to the baffle, so cut those to final size and hot-melt them into place.

The topmost ducts can be as much as 15″ long (the prototypes are 12″), which will tune the box to about 30Hz, while

**TABLE 1**
**PARTS LIST**

The following is a list of primary parts in the approximate order in which they are installed. All parts are ½″ plywood except as specified. All dimensions are nominal. Actual dimensions depend on the materials used and must be confirmed upon assembly.

| | |
|---|---|
| 1. Throat-horn sides (2) | 29″ × 7½″ |
| 2. Throat-horn divider | 8″ × 6″ |
| 3. Throat-horn sheathing (4) | 13½″ × 7″ (⅛″ plywood) |
| 4. Top, bottom | 29½″ × 29½″ |
| 5. Horn supports (2) | 29″ × 3¾″ |
| 6. Tweeter baffle | 29″ × 4″ |
| 7. Baffle | 21″ × 14¼″ |
| 8. Mouth-horn sheathing (2) | 29″ × 22″ |
| 9. Back braces (2) | 23″ × 5½″ |
| 10. Bend reflectors (2) | 29″ × 4½″ (halved 4″ ID PVC pipe) |
| 11. Side braces (4) | 20″ × 12½″ |
| 12. Back (4) | 31″ × 17″ (⅛″ plywood) |
| 13. Sides (2) | 30″ × 22″ |

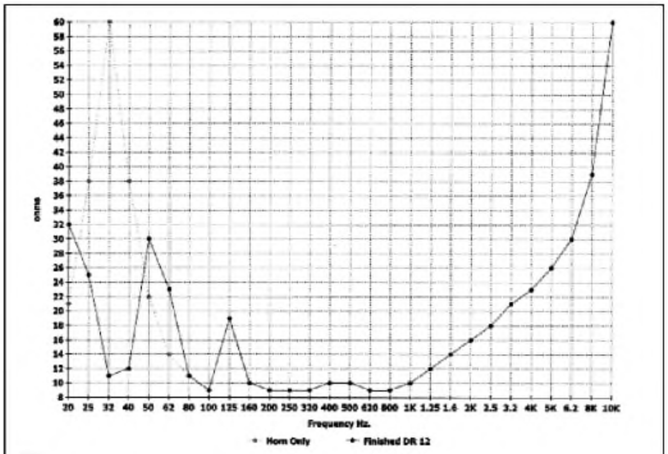


**FIGURE 6: Prototype measurement.**

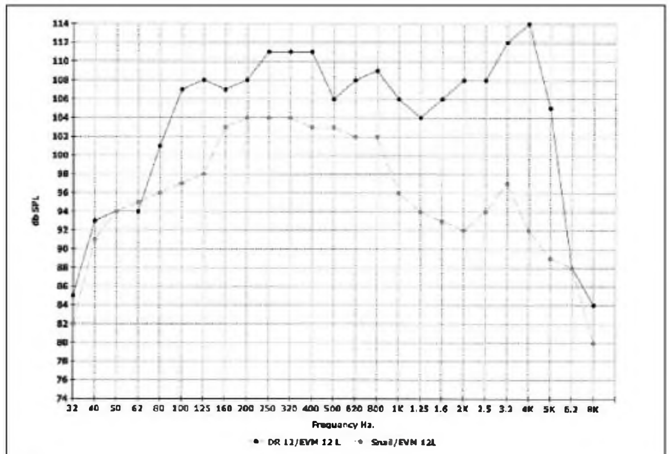


**FIGURE 8: Snail response.**

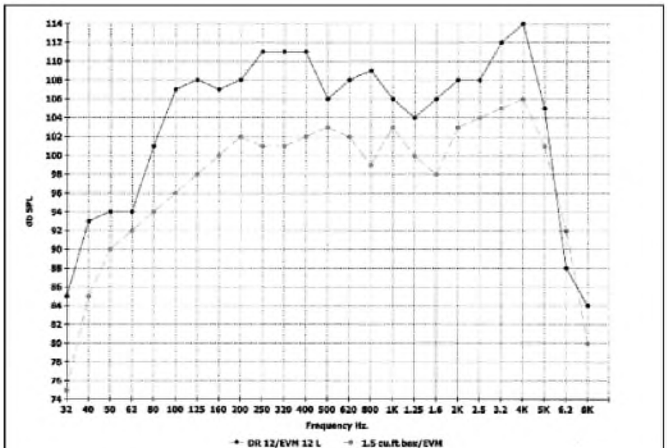


**FIGURE 7: DR12 response.**

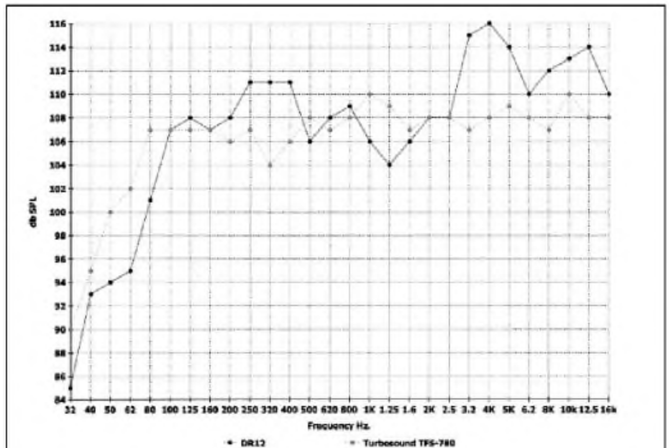


**FIGURE 9: DR12/Turbosound comparison.**

cutting them to 6″ will give an Fb of about 40Hz. Use whatever lengths you need, so that when you sweep-test the impedance there is now an impedance low point instead of a peak at the Fs(h). Make sure that the ducts, and the rear of the driver, are not blocked with any stuffing. Also note that if you ever need to remove the driver, the ducts nearest the porthole must be removed.

Your cabinet is now complete save for a finish, casters, and handles as desired. Paint is your cheapest and easiest finishing alternative. I prefer carpet, which is cheap and easy to apply and looks great. For a roadie-proof cab, use a laminate (such as Formica™) and aluminum edging.

## PERFORMANCE

So, what will this baby do? Looking at *Fig. 7*, average woofer sensitivity is 109dB (±4.5) from 100Hz to 5kHz. For comparison, the second trace on the chart is the same driver in a 1.5ft$^3$ closed box. Average SPL is only 100dB, and the curve, while not identical to the DR12, is no flatter. Almost shocking are the relative responses at 4kHz. The closed box has a peak here at 106dB. The reason for this is the EVM-12L driver. Like most extended-range drivers, it has a very light dust cover that acts as a midrange cone, giving superior mids until voice-coil inductance chokes off response.

In the past, this would have been a moot point for a folded horn. As shown in *Fig. 8*, my original Snail, using the same driver, rolled off above 800Hz. This was a vast improvement over previous folded-horn designs, where response would die as low as perhaps 400Hz, but it pales in comparison to the DR12, which maintains its 8dB advantage over the closed box right to the limit of the driver's range. This folded horn has the same bandwidth as a direct radiator.

Going back to Marty McFly's quest for loud, should you use this cab for guitar? I wouldn't. The 1.5ft$^3$ box with an EVM 12L has more than enough stage volume for those who'd rather not go deaf anytime soon. For keyboard? It would sound great, but is a lot bigger than you need to do the job unless you're playing stadiums. The DR10 is a better choice.

Bass? Maybe. While the DR10 is adequate for all but the largest gigs, the DR12 out-powers a commercial four ten/eighteen sub setup. In fact, it could replace two of those combinations, making it very attractive in both size and weight—not to mention cost, at only about $250 to $300 built as described. With its 5kHz bandwidth, it will work without tweeters, too, although you should add them if you desire an airy top end for slap and pop funk bass.

## LARGE-VENUE PA

Where this box really will shine is large-venue PA. With 300W input, each pair of these will comfortably put out (from 100Hz up) 138dB/1m. With two cabinets per stage side, 144dB is easily achievable from only one 600W/ch amp. While the 30 $in^3$ size may seem large, this is actually small within the genre of concert sound, and at 80 lbs, these almost qualify as featherweights.

For comparison, check out *Fig. 9* (which includes predicted response with two KSN-1177s). The dotted trace is the Turbosound TFS-780, the renowned "Flashlight." This is an active (quad-amped) four-way system, consisting of two cabinets of $13ft^3$ apiece. The low end is better than the DR12, but with twice the size (and a 21″ woofer) it should be. Above 100Hz, the DR12 is on average 2dB more sensitive. If you used two DR12s, giving you another 6dB broadband, you'd now equal the Turbosound below 100Hz, while above 100Hz, the DR12 advantage would be 8dB. That cuts your power-amp wattage requirements by at least 80%, and requires only one amp channel, not four.

That's not the only advantage of the DR12. If you wish to buy a TFS-780, you'll need to purchase it as part of a complete system, consisting of six cabinets, a power rack containing four amps and a controller unit, cables, software, and a 44-page manual. Good sound, but not exactly simple. And the cost? Let's just say it would be on par with another British export: the Rolls-Royce. Build your own DR12s and buy a tour bus with the savings.

After seeing what the DR10 and DR12 are capable of, those of you who have built Snail-series cabinets may question whether these designs have rendered your gear obsolete. The answer is not quite, since Snails are still better than commercial cabinets. On the other hand, if you demand only the best, you may prefer to scrap your Snails in favor of DRs. As few of you need anything as large or powerful as either the DR10 or the DR12, there will be three more smaller DR cabinets in this series, for 8″, 10″, and 12″ drivers. These are the same drivers used in the Snail series, so you'll only need to buy more plywood to make the upgrade.

On a personal note, a few of you have managed to get through to me on the phone to ask technical questions. I love speaking with readers, but the phone calls always seem to come at the worst times. If you do have a question on any of my projects, or even if you just want to chat, I have authorized *audioXpress* to give out my e-mail address to those of you who request it. My only caveat is that I may forward your question, and my answer, to *audioXpress* for publication. I'm looking forward to hearing from you.

Watch for my next project, the DR8. Thanks for your support, and remember—horns rule! ❖

Bill Fitzmaurice's Snail designs are detailed in his book, *Loudspeakers for Musicians* (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, custserv@audioXpress.com, $9.95).

# A Two-Channel Tube Compressor

Professional-sounding home recordings require a good compressor. This project features the superior sound of tubes, with a short signal path and versatile controls. **By Scott Swartz**

On rock or pop recordings, many tracks are typically compressed, including vocals, drums, and guitars, and often the entire mix is compressed when mixing to stereo. Since the entire mix may be routed through a compressor, the sound imparted by the compressor is very important. The warm sound quality imparted by tubes gives a tube-based compressor great potential.

## VARIOUS TOPOLOGIES

Many compressor topologies could be implemented with tubes. One possibility is to use a grid-controlled tube and a fixed resistor as a grid-controlled voltage divider. An example of this is shown in *Audio Anthology* Volume 2 as a volume expander, but it could be easily reconfigured as a compressor. The disadvantage to this topology is that it requires balanced stages and an output transformer to isolate the DC level shifts. This was more complexity than I wished to deal with.

Another possible topology is to use a variable-mu grid tube such as the 6L7G to vary the gain. I built a prototype using this, and found that the distortion was excessive, even when I kept the gain constant. Rickard Berglund's balanced 6ES8-based compressor in *Performer's Audio* 2/97 is another example of a variable-mu topology.

I chose to use an optoelectronic topology that utilizes a photoresistor controlled by a light source to instantaneously vary the gain. The light source can be a LED or an electroluminescent display, such as that used on the classic Teletronix LA-2 compressor. The LA-2 used the variable resistor as part of an attenuator; I chose to use the variable resistance as the feedback resistor in a negative feedback loop.

I did experiment with the LA-2 topology, but found that the distortion was a little high. By using the variable resistor in the feedback loop, distortion is generally lower and also decreases as the signal is compressed. The concept of using a variable-feedback resistor was published in *GA* 2/92 in Eric Barbour's article, "Vacuum Tube AC Feedback Amplifiers." Note that any optical topology typically gives a fairly slow RMS-type compression that will not compress very short peaks, but this type of compression usually sounds the most natural, anyway.

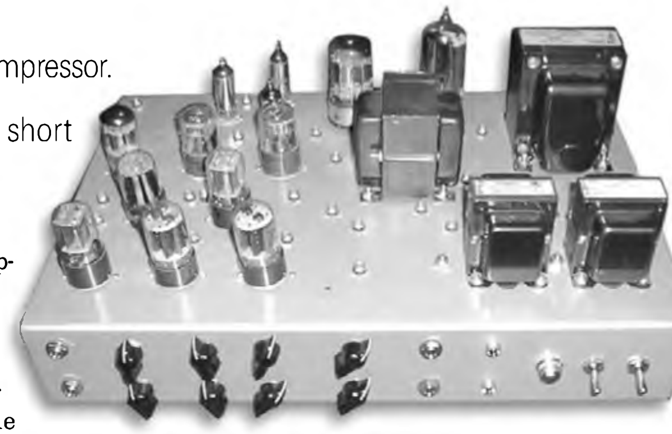


**PHOTO 1: Exterior view of the compressor.**

## CIRCUIT DESIGN

The basic circuit consists of a variable-

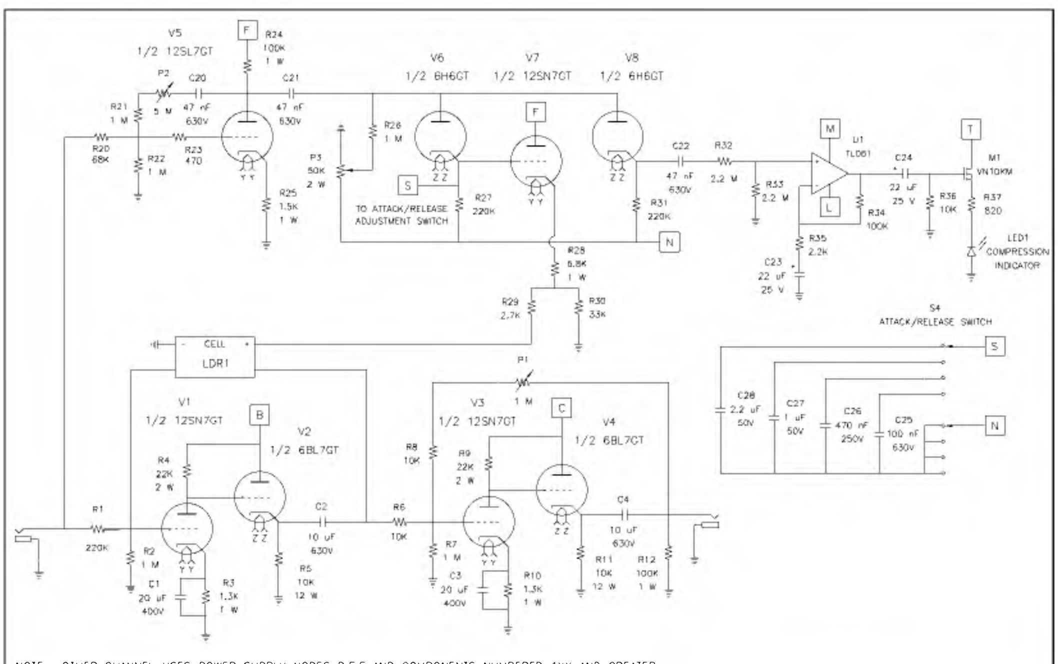



**FIGURE 1: Compressor circuit.**

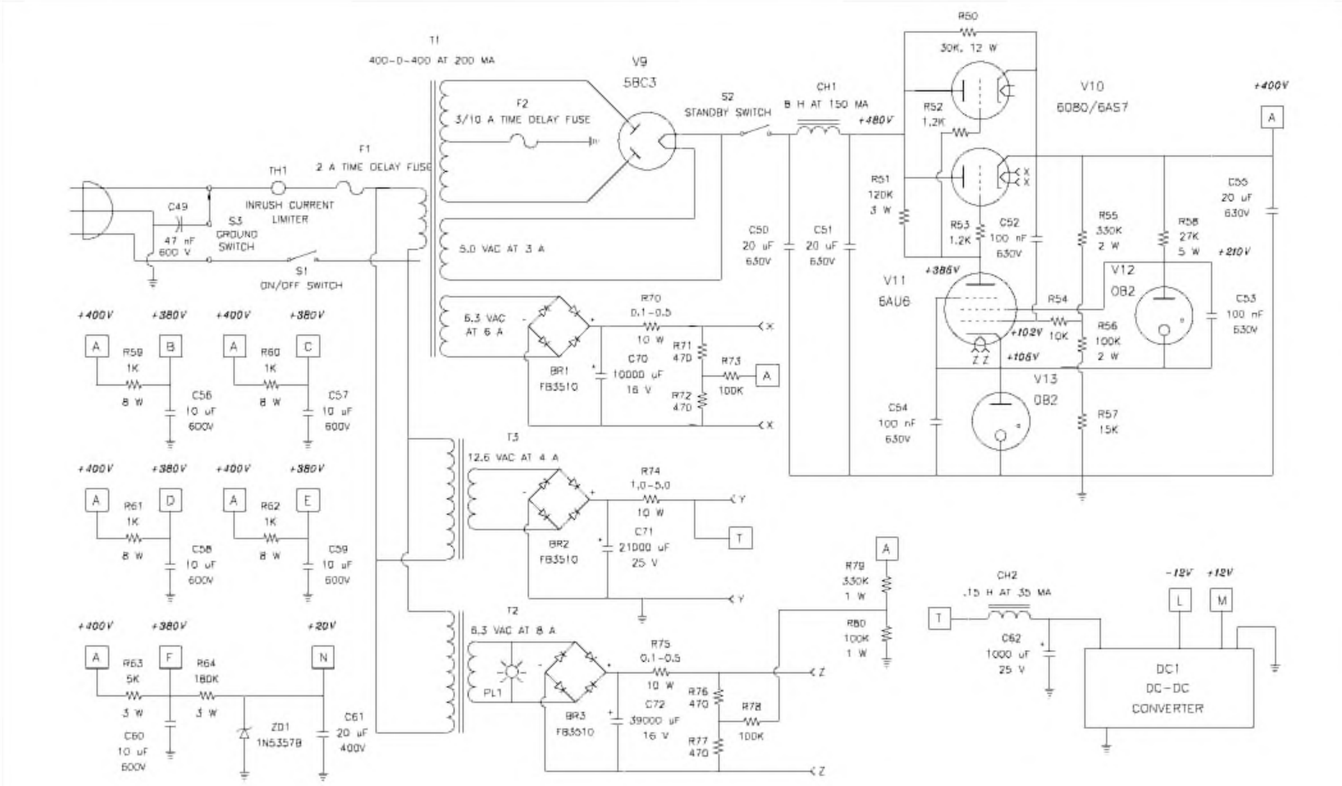



**FIGURE 2: Power supply.**

gain input stage, a side-chain circuit that controls the gain of the input stage, and an output amplifier (*Fig. 1*). The side chain varies the instantaneous gain of the input stage, reducing the gain of signals over a user-determined threshold by a user-determined ratio. For instance, if you set the threshold at 0dB and the ratio for 2:1, an input level of +2dB will be compressed by 1dB, and the actual output level will be +1dB. The output amplifier sets the output to the desired level after compression. The compressor unit is shown in *Photo 1.*

The input stage consists of a 12SN7GT direct-coupled to a 6BL7 cathode follower. A negative-feedback loop that is instantaneously adjusted by the side chain controls the stage gain. The input stage typically operates at a net gain of less than 1, because of the signal loss in the input-resistor network and the light-dependent-resistor (LDR) feedback. The cathode follower is necessary to lower the stage's output impedance so the fairly low value of feedback resistance does not cause distortion. The relatively high series resistance between the input and the grid of the input stage is necessary to isolate the side chain from the input-stage feedback.

**TABLE 1**
**POWER REQUIREMENTS OF EACH STAGE**

| STAGE | TUBE | VOLTAGE | CURRENT (TOTAL) |
|---|---|---|---|
| Input gain stage | 12SN7/6BL7 | +380V | 60mA |
| Side chain amplifier | 12SL7 | +380V | 4mA |
| Cathode follower | 12SN7 | +380V | 8mA |
| Output amplifier | 12SN7/6BL7 | +380V | 60mA |

The side chain consists of a side-chain amplifier, a diode, a cathode follower, and a LDR, which is an assembly of a photoresistor and a LED. A detailed description of each stage follows.

The side-chain amplifier, a 12SL7, is a simple tube gain stage that samples the input signal and amplifies it. The gain of the side-chain amplifier is varied by a potentiometer-adjustable negative-feedback loop, which provides the compression ratio control.

The output of the side-chain amplifier is coupled to the side-chain diode, a 6H6GT that rectifies the signal into a varying DC level. The cathode of the diode is returned to a +20V DC source. You can adjust the DC bias of the diode's plate between 0–20V DC, which provides the threshold control. The capacitance value across the diode cathode resistor can be switched to one of four values, which varies the attack-and-release speed. The cathode of the diode is direct-coupled to the cathode follower.

## CATHODE-FOLLOWER FUNCTIONS

The cathode follower, a 12SN7, functions as a variable-current source for the LDR, where the current is controlled by the DC level on the cathode-follower grid. This varying current controls the intensity of the LED inside the LDR. The light intensity inversely varies the resistance of the photoresistor, which is the feedback resistor for the input stage. Therefore, when a signal causes the DC level on the cathode of the side-chain diode to rise, the current through the cathode follower increases, which in turn decreases the value of the input-stage feedback resistor, thus reducing the input-stage gain.

The total standing current through the 12SN7 is approximately 4mA,


**ABOUT THE AUTHOR**

Scott Swartz received a BS in mechanical engineering from the University of Missouri. He has been employed as a mechanical engineer in positions including product design and facility engineering. Scott has been a musician since the age of 12, playing the guitar, drums, keyboards, and several other instruments. He is actively involved in home recording and building tube-based devices for performance, recording, and playback.

which corresponds to a feedback resistor of 70k. If desired, you can increase the maximum compression ratio by lowering the +20V DC source to +10V DC, which increases the current change for a given voltage change on the cathode-follower grid. This would also require adjusting the value of the current-division network resistors R28, R29, and R30, so that the LDR standing current remains 3.25mA. You may also need to adjust the value of the current-division network resistors to account for LDR tolerance.

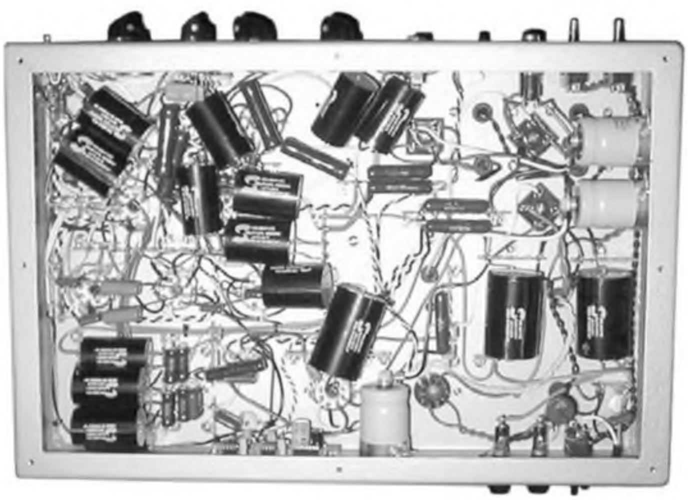


**PHOTO 2: View of the interior.**

During the construction, I discovered a quirk of LDRs that influenced the LDR standing current just described. I originally set out to use a higher range of feedback resistance (about 250k) so that an output amplifier would not be required to bring the output level back up to –10dBm. Intuitively, this requires only reducing the standing current through the LDR.

However, I found that at low LDR currents, the LDR feedback was actually adding distortion! The LDR feedback reduced the gain as negative feedback normally does, but the distortion was increased by as much as 3% THD. As the LDR resistance diminished, the added distortion decreased, and finally, at around 100k, the LDR resistance began to reduce the distortion. I therefore selected a feedback value below 100k.

I found a possible answer to this puzzle on the website of the LDR's manufacturer (www.perkinelmer.com/opto-110/vac trols.htm). The distortion through an LDR is very low at 1V RMS, but at 10V RMS it will be 2–3% THD. This behavior must have something to do with the high distortion I saw with the high feedback resistance. This website has extensive information on all aspects of LDRs, including performance specifications such as resistance range, response time, and so on.

## OUTPUT AMPLIFIER

The output amplifier consists of a 12SN7GT direct-coupled to a 6BL7 cathode follower, which allows the compressor to easily drive impedances of 5k or higher. A potentiometer-adjustable negative-feedback loop controls the output level. The output amplifier's main purpose is to restore the compressed output signal to the level of the input signal, but the range of gain is wide enough to provide a higher output than input level. This can be useful in matching –10dBm to +4dBm equipment.

The compression-indication LED is controlled by a 6H6GT that is in parallel with the 6H6GT side-chain diode. The anode and cathode of both 6H6GTs are at the same quiescent DC potentials. When the side-chain diode 6H6GT is conducting and changing the DC level of its cathode, the indicator 6H6GT passes a rectified signal to the indicator op amp, which is RC-coupled to the gate of a MOSFET. The indicator LED is in series between the MOSFET source and the ground. The gain of the op amp is set so that as soon as the 6H6GT starts to conduct, the LED lights up.

There may be a more elegant way to utilize solid-state technology to provide the indication than I have used; I started in electronics using only tubes, and this is actually the first time I have used solid-state components. One problem with the circuit shown is that the indication circuits are slightly interactive, i.e., when only one channel is in use, the indicator for the other dimly lights. I worked around this by installing input jacks that incorporate an isolated break circuit that grounds the input of the op amp for a given channel when that channel is not in use.

This interaction probably happens because the power supplies for the two indicator circuits are not decoupled; I did not have room to do this because I added the indicators after the enclosure was complete. The interaction does not occur in the audio sections of the compressor because the audio stages are decoupled.

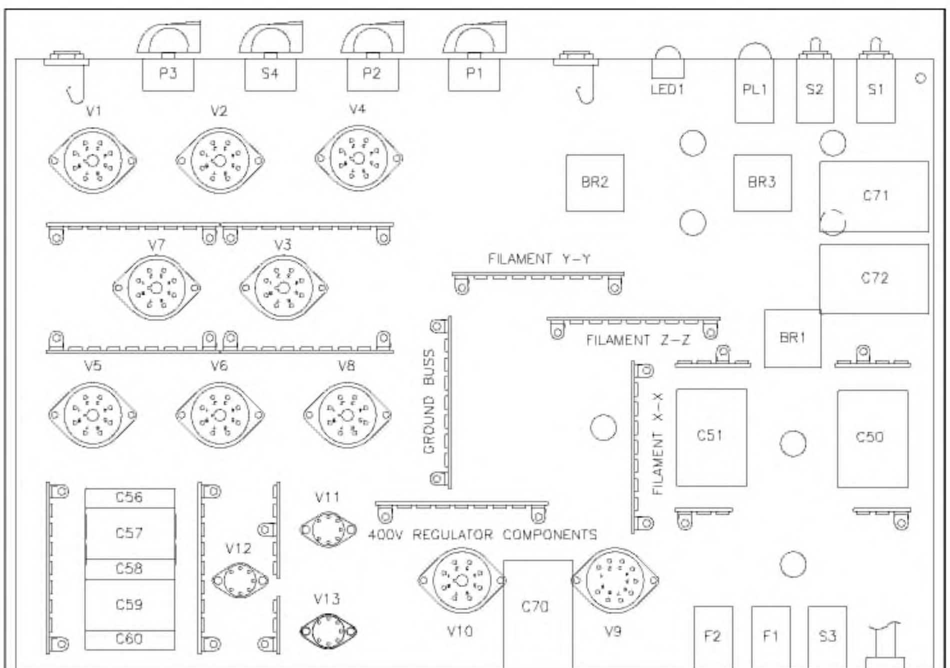



**FIGURE 3: Chassis interior layout.**

You may be wondering why I have not mentioned any provision for input-level adjustment in this circuit description. I prefer to keep the signal path as short as possible, so I avoid input amplification or attenuation. The circuit as shown is optimized only for –10dBm (1V pk-pk) inputs. Levels higher or lower will shift the range of adjustment for the ratio and threshold controls, possibly making the desired compression settings unattainable. Since a compressor is typically inserted after a mike preamp—or between the mixer and the recording device, where the levels will typically be –10dBm—the lack of input-level adjustment has not been a problem.

The above discussion is for one channel of the compressor, as is the circuit schematic. The tubes used are twin triodes/diodes with half of each tube used for a given channel. I built the prototype with two independent channels, but you could build stereo with one set of controls by using stacked potentiometers.

## POWER SUPPLY

The power supply (*Fig. 2*) is conventional in design. The power transformer supplies a pi filter, and the raw voltage is adjusted to the desired value using a vacuum-tube regulator. Because the intended use for the compressor is recording, low hum and noise are very desirable; active regulation is the surest way to eliminate power-supply hum and voltage disturbances such as spikes. The compressor circuit would work fine with an unregulated supply if you wish, but then hum may be more audible. The power requirements of each stage are summarized in *Table 1*.

The power supply uses a Hammond 278X transformer rated 400V-0V-400V at 200mA, which feeds a 5BC3 full-wave rectifier and a pi-filter section. The raw supply gives around +480V DC at the design current. A regulator using a 6080 pass tube and a 6AU6 error amplifier is used to provide +400V DC. A 0B2 voltage reference tube is used as the error-amplifier cathode-voltage reference and also to regulate the 6AU6 screen voltage.

The design of this regulator is similar to Douglas McCann's Purist Tube Regulator described in great detail in *GA* 3/94. See also "Vacuum-Tube Regulator Design," by Denzil Danner in *GA* 3/93. All filter capacitors are metallized polypropylene and are modestly sized to provide good transient response.

**TABLE 2**
**FILAMENT INFORMATION**

| TUBE | FILAMENT VOLTAGE | FILAMENT CURRENT | CATHODE VOLTAGE | FILAMENT BIAS | FILAMENT TRANSFORMER |
|---|---|---|---|---|---|
| 6080 | 6.3V | 2.5A | +400V | +400V | T1 |
| 6BL7GT (2) | 6.3V | 3.0A | +225V | +100V | T2 |
| 6H6GT (2) | 6.3V | 0.6A | +20V | +100V | T2 |
| 6AU6 | 12.6V | 0.3A | +105V | +100V | T2 |
| 12SN7GT (3) | 12.6V | 0.9A | +8V/+35V | 0V | T3 |
| 12SL7GT | 12.6V | 0.15A | +2V | 0V | T3 |

The compressor is controlled by on/off and standby switches, similar to the design used on many guitar amplifiers. To power up the compressor, move the on/off switch to the on position, and after 30 seconds move the standby switch to the on position, which applies the B+ to the compressor circuit. This process could be automated with the use of relays.

You should never apply the B+ until the filaments are warmed up, since this causes cathode stripping, resulting in shortened tube life. It can also cause the pass tube of a vacuum-tube regulator to arc. A resistor connected between the plate and cathode of the pass tube will solve the arcing problem, since it provides a "bypass" for the current while the tube warms up. It does not stop the cathode stripping, however.

The grounding arrangement I used on the prototype is a centrally located ground bus, with the chassis connected to it by a single wire. I grouped circuit grounds together by stage, and the groups run back to the bus with a single wire. This technique seems to work as well as true star grounding, but it is easier to implement. See *Photo 2* for an interior view.

## FILAMENTS

All filament supplies are DC, using a bridge rectifier, a filter cap sized for approximately 10,000µF per filament amp, and a dropping resistor. I find that adding these few components is a small price to pay for eliminating all concerns regarding filament-related hum. All filament transformers are sized at no more than 70% of their rated current to allow for the current peaks caused by the rectification to DC.

The 6.3V winding of power transformer T1 is biased to +400V DC, and it supplies the 6080 filament. Transformer T2, rated 6.3V AC at 8A, supplies the two 6BL7s, the 6AU6, and the two 6H6GTs, and is biased at +100V DC to keep the heater-cathode voltage difference at an acceptable value. The three 12SN7s and the one 12SL7 are run off filament transformer T3, which is rated at 12.6V AC at 4A. This information is summarized in *Table 2*.

You could build the compressor entirely with 6V heater tubes, but since a

### TABLE 3
### PARTS LIST

| RESISTORS | VALUE | VOLTAGE | TYPE | SUPPLIER |
|---|---|---|---|---|
| R1, R101 | 220k | 1W | carbon composition (CC) | Surplus |
| R2, R7,R102, R107 | 1M | 1W | CC | Surplus |
| R3, R10, R103, R110 | 1.3k | 1W | CC | Surplus |
| R4, R9, R104, R109 | 22k | 2W | CC | Surplus |
| R5, R11, R105, R111 | 10k | 12W | wirewound | New |
| R6, R8,R106, R108 | 10k | 1W | CC | Surplus |
| R12, R112 | 100k | 1W | carbon composition | Surplus |
| R20, R120 | 68k | 1W | carbon composition | Surplus |
| R21, R22, R26, R121, R122, R126 | 1M | ½W | metal film (MF) | MSR |
| R23, R123 | 470 | ½W | MF | MSR |
| R24, R73, R78, R80, R124 | 100k | 1W | carbon film (CF) | MSR |
| R25, R125 | 1.5k | ½W | MF | MSR |
| R27, R31, R127, R131 | 220k | ½W | MF | MSR |
| R28, R128 | 6.8k | 1W | CF | MSR |
| R29, R129 | 2.7k | ½W | MF | MSR |
| R30, R131 | 33k | ½W | MF | MSR |
| R32, R33, R132, R133 | 2.2M | ½W | MF | MSR |
| R34, R134 | 100k | ½W | MF | MSR |
| R35, R135 | 2.2k | ½W | MF | MSR |
| R36, R136 | 10k | ½W | MF | MSR |
| R37, R137 | 820 | ½W | MF | MSR |
| R50 | 30k | 12W | wirewound | New |
| R51 | 120k | 3W | metal oxide (MO) | MSR |
| R52, R53 | 1.2k | ½W | MF | MS |
| R54 | 10k | ½W | MF | MSR |
| R55 | 330k | 2W | MO | MSR |
| R56 | 100k | 2W | MO | MSR |
| R57 | 15k | ½W | MF | MSR |
| R58 | 27k | 5W | wirewound | New |
| R59–62 | 1k | 5W | wirewound | New |
| R63 | 5k | 3W | wirewound | New |
| R64 | 180k | 3W | MO | New |
| R70, R75 | 0.1–0.5 | 10W | wirewound | New |
| R71, R72, R76, R77 | 470 | ½W | MF | MSR |
| R74 | 1.0–5.0 | 10W | wirewound | New |
| R79 | 330k | 1W | CF | MSR |

| ITEM | PART NUMBER | DESCRIPTION | MANUFACTURER | SUPPLIER |
|---|---|---|---|---|
| V1, V3, V7 | 12SN7GT | twin triode | as desired | as desired |
| V2, V4 | 6BL7GT | twin triode | as desired | as desired |
| V5 | 12SL7GT | twin triode | as desired | as desired |
| V6, V8 | 6H6GT | dual diode | as desired | as desired |
| V9 | 5BC3 | full-wave rectifier | as desired | as desired |
| V10 | 6080 | twin triode | as desired | as desired |
| V11 | 6AU6 | pentode | as desired | as desired |
| V12, V13 | 0B2 | VR tube, 105V | as desired | as desired |
| LDR1, LDR101 | VTL5C6 | analog optoisolator dark R = 100M | Vactrol | Newark |
| BR1–BR3 | FB3510 | bridge rectifier, 35A, 1000 PIV | Fagor, IR | New, MSR |
| ZD1 | 1N5357B | zener diode, 5W Vz = 20V | Fagor, Motorola | New |
| U1, U101 | TL061 | JFET input op amp | as desired | as desired |
| M1, M101 | VN10KM | MOSFET, 50V 200mA | as desired | as desired |
| LED1, LED101 | N/A | LED, VF = 2V | as desired | as desired |
| PL1 | N/A | 6V pilot light | as desired | as desired |
| S1, S2 | N/A | switch, SPST, rated 10A at 250V AC | as desired | as desired |
| S3 | N/A | switch, SPDT, rated 10A at 250V AC | as desired | as desired |
| S4, S104 | N/A | switch, 3P4T rotary | as desired | as desired |
| TH1 | CL101 | inrush current limiter, 16A | Keystone | MSR |
| F1 | MDL-2 | fuse, 2A time delay | Bussman | MSR, New |
| F2 | MDL-3/10 | fuse, 3/10A time delay | Bussman | MSR, New |
| P1, P101 | N/A | 1M audio ½W potentiometer | Alpha | MSR |
| P2, P102 | N/A | 5M audio ½W potentiometer | Alpha | MSR |
| P3, P103 | N/A | 50k linear 2W potentiometer | as desired | Surplus |
| T1 | 278X | power trans., 400V-0V-400V | Hammond | AES, AI |
| T2 | 167N12 | filament trans., 12.6V at 4A | Hammond | AES, AI |
| T3 | 167R6 | filament trans., 6.3V at 8A | Hammond | AES, AI |
| CH1 | 193D | choke, 8H at 150mA | Hammond | AES, AI |
| CH2 | 434-03-154J | choke, .15H at 36mA | Xicon | MSR |
| DC1 | NMH1212D | DC-DC converter ±12V at 67mA | Newport | MSR |

continued on next page

| CAPACITORS | VALUE | WVDC | TYPE | MANUFACTURER | SUPPLIER |
|---|---|---|---|---|---|
| C1, C3, C61, C101, C103 | 20µF | 400V | metallized polypropylene | Solen | AI, AES |
| C2, C4, C56–60, C102, C104 | 10µF | 630V | metallized polypropylene | Solen | AI, AES |
| C20–C22, C49, C120–C122 | 47nF | 630V | metallized polypropylene | Xicon | MSR |
| C23, C24, C123, C124 | 22µF | 25V | electrolytic | Xicon | MSR |
| C25, C125 | 100nF | 630V | metallized polyester | Xicon | MSR |
| C26, C126 | 470nF | 250V | metallized polyester | Xicon | MSR |
| C27, C127 | 1µF | 50V | electrolytic | Xicon | MSR |
| C28, C128 | 2.2µF | 50V | electrolytic | Xicon | MSR |
| C50, C51, C55 | 20µF | 630V | metallized polyproplene | Solen | AI, AES |
| C52–C54 | 100nF | 630V | metallized polyproplene | Xicon | MSR |
| C62 | 1000µF | 25V | electrolytic | Xicon | MSR |
| C70 | 10000µF | 16V | electrolytic | Xicon | MSR |
| C71 | 21000µF | 25V | electrolytic | Cornell-Dubilier | New |
| C72 | 39000µF | 16V | electrolytic | Cornell-Dubilier | New |

**KEY TO SUPPLIER ABBREVIATIONS**

| | |
|---|---|
| AES | Antique Electronic Supply |
| AI | Angela Instruments |
| MSR | Mouser Electronics |
| New | Newark Electronics |

separate filament transformer is required for the SN7s and the SL7s (because of the 0V bias), I chose to use 12SN7s and 12SL7s since they are less than half the price of 6SN7s.

## COMPONENT CHOICES

All resistors in the compressor circuit are carbon composition, except for the 6BL7 cathode resistors, which are wirewounds because of the high dissipation required. I find the carbon comps to have a much smoother sound, especially on high-frequency sounds such as cymbals. You should experiment to find which components suit your ear. Resistors in the power supply are wirewound, or carbon, or metal film, depending on the wattage required.

Possible suppliers for the major components are indicated in the parts list (*Table 3*). The list of suppliers is not exhaustive. You can purchase many of the components listed (or close substitutes) at surplus stores. Hardware items such as jacks, power cords, tube sockets, and so on, are not shown.

## CONSTRUCTION

You'll find it efficient to construct the sections of the compressor in the correct sequence and carefully test each section before moving to the next, since then you won't have to check all the stages to find a wiring error. I recommend the following order of construction.

1. Prepare a detailed layout of component placement. The layout I used on the prototype is shown in *Figs. 3* and *4*. *Figure 5* shows the front-panel layout. Change this if you like, but be sure to adhere to the following construction guidelines:

A. Mount all power and filament transformers as far as possible from the input circuitry.

B. The laminations of the filament

and power transformers should be perpendicular to the laminations of the filter choke.

C. The center-to-center spacing of the preamp and regulator tubes should be 2½″ minimum.

The dimensions of the chassis used on the prototype are 12″ deep × 18″ wide × 3″ high, constructed of 16 GA steel with fully welded corners. The bottom cover should be perforated to promote cooling of the internals; the 6BL7GT cathode-follower resistors in particular put out a lot of heat.

2. Drill or punch all holes, including temporary mounting of transformers.

3. Apply desired finish to chassis.

4. Mount the power and filament transformers. All wires passing through the chassis should have grommets to protect the insulation. Wire the 120V AC input components such as line cord, ground-select switch, fuse, and so on.

5. Wire the filament DC supplies. It will be necessary to try different voltage-dropping resistors to get the required filament voltage. For 6.3V DC supplies, the resistor is in the range of 0.1–0.5Ω, and for the 12.6V DC supply, it is in the range of 1.0–5.0Ω. Be sure the 470Ω filament-bias resistors are in place where required.

6. The next step is to wire the power supply. Build and test the raw voltage supply before starting on the regulator. With no current draw, the supply should measure 560V DC.

7. Begin wiring the +400V DC regulator. Do not delete the resistor connected from the plate to the cathode of the pass tube, which prevents arcing at startup. You should adjust the small-value resistor R57 if necessary to give the proper output voltage. Test the regulator at various current draws to verify regulation; this requires high-wattage wirewound resistors. A 3k, 100W resistor will test the regulator at 130mA.

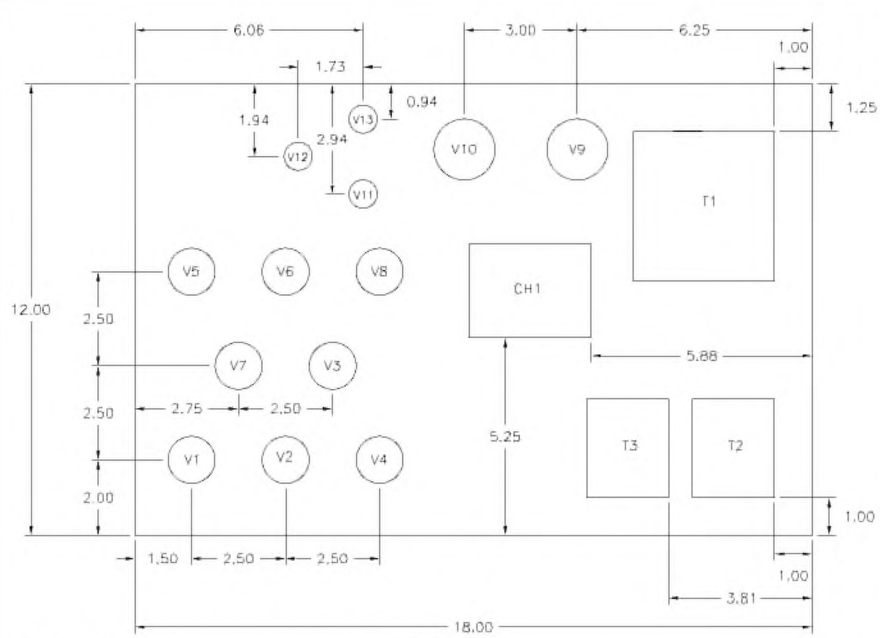


**FIGURE 4: Chassis exterior layout (top view).**

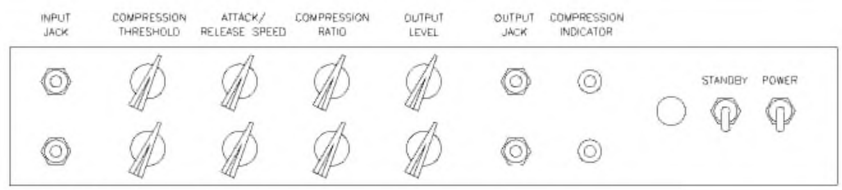


**FIGURE 5: Front panel layout.**

8. Wire the stage decoupling capacitors and resistors.

9. Wire the +20V DC supply, which consists of a resistor, zener, and capacitor.

With the power supply complete, you can now wire the compressor circuit. Note that the schematic shows only one channel, and that the second channel uses the other halves of the dual triodes/diodes. The construction should proceed as described subsequently, because space is limited due to the large size of some of the components, especially the 20μF film cathode-bypass capacitors.

10. You should first wire the input jacks, since this area will be inaccessible after you wire the input stage. The input-signal division resistors attach to a terminal strip and the socket for V1. Wire the resistors that mount to the terminal strip back to the side-chain amplifier.

11. You should next wire the side-chain amplifiers and the ratio controls. The ratio pots are part of the feedback loop of the side-chain amplifier.

12. Wire the threshold pots and run overlength wires to the side-chain diode terminal strip, the ground bus, and the +20V DC node.

13. Wire the attack/release speed-adjustment switches on the bench with overlength leads that you will later connect to the side-chain diode and the +20V DC node. Mount the switches to the chassis, then cut the overlength leads to size and solder them.

14. Wire the side-chain and indicator diodes. The solid-state indicator components, which I mounted on perfboard to the back panel of the chassis, should be wired next.

15. Now wire the cathode followers, LDR cells (the LED section), and the as-

## SOURCES

Angela Instruments
10830 Guilford Rd., Suite 309
Annapolis Junction, MD 20701
301-725-0451
www.angela.com

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
602-820-5411
www.tubesandmore.com

Mouser Electronics
800-346-6873
www.mouser.com

Newark Electronics
800-463-9275
www.newark.com

sociated resistors.

16. At this point, you should check the operation of the side chains. First, the LDR quiescent resistance needs to be checked (quiescent meaning no AC input, but the compressor powered up). You must do this test before you connect the LDR resistance element to the feedback loop. This resistance should measure 70–100k; you can vary it by changing the values of R28, R29, and R30. R28 will provide large changes to the current, and R30 is used for fine adjustment.

The schematic is shown with the three-resistor network, so the LDR current can be less than the total at a given bias point for the cathode follower. Using a 1V pk-pk sine wave and an oscilloscope, check the output of the side-chain amplifier and see that the ratio control varies the amplitude as you sweep the control. With the ratio control set at its midpoint, check the threshold control by setting it at its midpoint and increasing the amplitude of the sine wave, starting from zero.

The DC voltage on the grid of the cathode follower should start to increase when the sine wave reaches approximately 1V pk-pk, indicating that compression is beginning. Continue raising the amplitude of the sine wave and verify that the DC grid voltage continues to rise. This rising voltage indicates more current through the LDR, and therefore a lower value of feedback resistor.

17. After verifying the operation of the side chains, wire the input stages and output amplifiers.

## POSSIBLE MODIFICATIONS

One possible modification would be to use mu-follower gain stages instead of the gain stage direct-coupled to a cathode follower. I tried this temporarily on one channel, and after comparing the two, I decided to go with the warmer sound of the gain stage directly coupled to the cathode follower.

Another possible modification might be to use more solid-state components in the side chain. For instance, the current source for the LDRs could be a FET, and you could use germanium diodes instead of the 6H6GTs. I suspect this would not impact the sound, since the LDRs isolate the side chain and signal path, and it would save space.

Other possible modifications include different signal-path tubes, a simpler nonregulated power supply, or different component choices. With regard to the tubes, possible alternatives for the SN7s could be any of the 12A_7 types such as 12AU7, 12AX7, 6CG7, or any medium- or high-mu dual triode. You could also replace the 6BL7GTs with a 5687, a 7119, or even a higher plate-resistance tube such as the 12AU7 if you can accept a slightly higher output impedance. Another possibility is dual dissimilar triodes such as 6EA7, 6DN7, 6FD7, and so on.

## THE SOUND

The sound of the compression is very fat, with a lot of tube warmth. The peaks are reduced without changing the character and frequency balance of the sound, which means an acoustic guitar sounds like an acoustic guitar, for instance. The compression is very unobtrusive, but will be easily visible on the recorder level meters. Note that the range of adjustment on this design is not as great as on a typical solid-state VCA compressor, so it is not well suited for special-effects compression, but for typical compressor applications, there is plenty of range. Noise and hum are very low; in fact, this compressor is much quieter than a dbx 266 compressor that I own.

## SUMMARY

When I started this project, I intended to design a device that would combine the superior sound of tubes, a short signal path, and parameter control as versatile as that of a solid-state VCA compressor. This design certainly fulfills the sound objective, with only a slight compromise on parameter control. I am very satisfied with this design and use it regularly when recording.

I have tried to explain each circuit building block thoroughly so that if you are interested in building a similar device, you can use the design in its entirety or use only a portion of the blocks. There is much room for further experimentation on this subject. ❖

# Digital Impedance Measurement

Armed with a PC and the right software, this author demonstrates how you can determine the all-important impedance measurement to design a loudspeaker system. **By Oliver H. McDaniel**

The impedance of a loudspeaker is an important quantity to know in the design of loudspeaker systems. You can compute the Thiele/Small parameters from suitable impedance measurements. Loudspeaker manufacturers often supply impedance curves and Thiele/Small parameters, but the curves are often poorly printed and difficult or impossible to read accurately, and the actual data for a given loudspeaker can vary substantially from the published data. Impedance measurements for vented-box systems are also very useful in verifying box tuning.

The concept of impedance is clearly explained in a recent article in *Speaker Builder* by Dick Pierce[1], and he presents a precise method for manually measuring impedance. He also mentions digital methods, but the software and equipment for these has, until recently, been quite expensive. Dedicated instruments for the required fast-Fourier-transform (FFT) measurements typically cost $20,000 or more.

The development of the soundcard and inexpensive (sometimes free) software for personal computers provides a drastic reduction in the cost of doing digital-audio signal processing. The soundcard is essentially a 16-bit, dual-channel, analog-to-digital (AD) and digital-to-analog (DA) converter with digital antialiasing filters. The accuracy of these cards is remarkable, as I will demonstrate later.

**ABOUT THE AUTHOR**

Oliver McDaniel has a PhD in Engineering Acoustics. He taught and did research in underwater acoustics and noise control for 30 years, until his retirement in 1995. He has taught courses in experimental methods in acoustics, noise control engineering, environmental noise, and fundamentals of acoustics.

## THE SOFTWARE

The most important software used in this study is the FFT analyzer, which allows you to determine the frequency content of a sampled time signal. The upper frequency limit, $f_{MAX}$, of the analysis is one half of the sample rate of the time signal. The frequency resolution is $f_{MAX}$ divided by the FFT size. For example, a time signal sampled at 44.1kHz with a 4096-point FFT would have a frequency resolution of 5.38Hz.

The software used in this study to perform the FFT analysis is AtSpec Pro[2] version 2, authored by Paavo Jumppanen. AtSpec will perform dual-channel FFT analysis with resolutions from 64 to 32,768 frequency points at soundcard sampling rates of 44.1, 22.05, and 11.025kHz. It will analyze .wav files (a digital-audio recording format) at any sampling rate. It will compute the transfer function required for impedance measurements and all the other digital signal-processing functions available on the expensive dedicated instruments.

The cost of the software is under $100 and it runs on Windows 95, 98, NT, and 2000. An earlier version (1.51) will run with Windows 3.1, but is limited to 2,048 frequency points. Data is exported from AtSpec as ASCII text files for further computations and graphing. An AtSpec screen shot is shown in *Fig. 1*.

I performed the computations and graphics for this article with Mathsoft's Axum 5.0, but they could easily be done with less expensive software such as Microsoft Excel or Lotus 1-2-3. Other useful software for audio measurements are Wavetools[3], GoldWave[4], and Cool Edit.[5] Wavetools contains a signal generator (periodic and random noise), a dual-channel oscilloscope, a level meter, and a single-channel FFT analyzer (good for a quick look at spectra), all operating with the soundcard.

GoldWave records and plays large .wav samples of stereo audio signals of preset length, and also does lowpass, highpass, and bandpass digital filtering of .wav recordings. Cool Edit generates

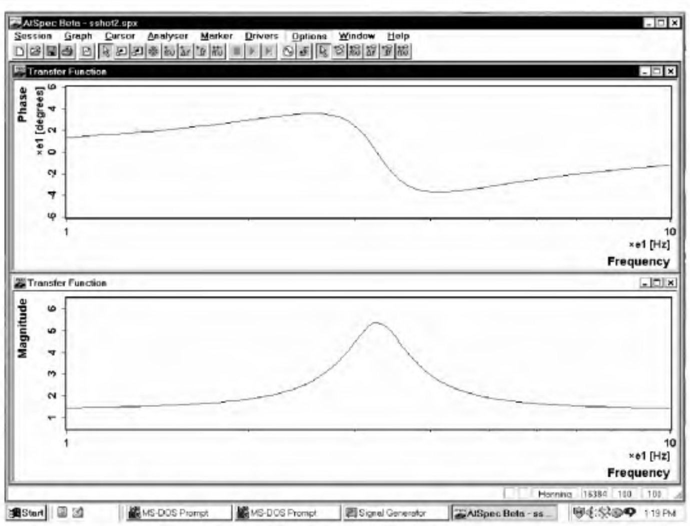



**FIGURE 1: Screen shot of AtSpec transfer function.**

B-1740-1

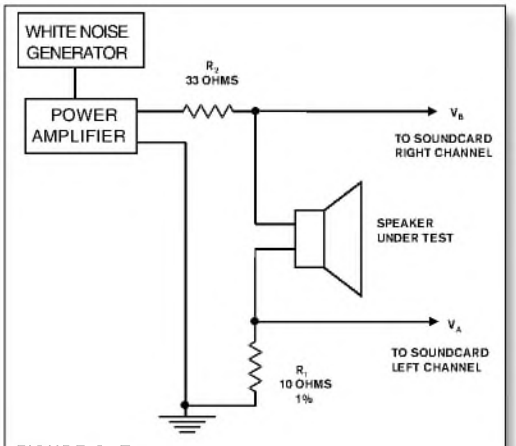



**FIGURE 2: Test setup.** B-1740-2

periodic and random noise .wav recordings and also does downsampling of .wav files. Demonstration versions of GoldWave and Cool Edit are available free from their websites, and they perform all of the mentioned functions.

## THE METHOD

The test setup for performing the measurements is shown in *Fig. 2.* The white-noise generator used here is actually the soundcard line output using the Wavetools' signal generator to produce the random noise. This software develops random noise by looping (continuously replaying) noise recorded on .wav files. It will operate with any .wav file in its subdirectory, but the setup will work only if the .wav file you select is at the same sample rate as the AtSpec sample-rate setting.

The white-noise .wav file supplied with Wavetools is short, at a 44.1kHz sample rate. I replaced this with much longer white-noise .wav files generated with Cool Edit at 44.1, 22.05, and 11.025kHz sample rates. These larger .wav files produce more statistically correct white noise, resulting in smooth FFT data with less averaging than with short .wav files. To use a soundcard in this way with both the line inputs and outputs operating, you must set it for full duplex operation.

The AtSpec channel convention is that signal A is the left channel of the soundcard and signal B the right. The voltages $V_A$ and $V_B$ should be set to less than 1V RMS to prevent clipping of the soundcard line input. You can monitor these voltages with an AC voltmeter, the VU-meter function of Cool Wave (good for all sampling rates), or Wavetools' meter (good only at 44.1kHz). You must exit these programs before running AtSpec or any other soundcard input programs, since soundcard input (or output) functions cannot be shared by any programs used here.

The voltage across $R_1$ ($V_A$) is proportional to the current supplied to the speaker under test, and the voltage $V_B$ is proportional to the current plus the voltage applied to the speaker. The resistor $R_2$ supplies the speaker with a relatively constant current that increases the current signal ($V_A$) around resonance, improving the signal-to-noise ratio of the measurement. It also provides short-circuit protection for the power amplifier. The resistors used here are 10W wirewound with 1% tolerance. The value of $R_2$ is not critical, and any power resistor between 20 and 50Ω will work well; however, the tolerance of $R_1$ affects the accuracy of the impedance results.

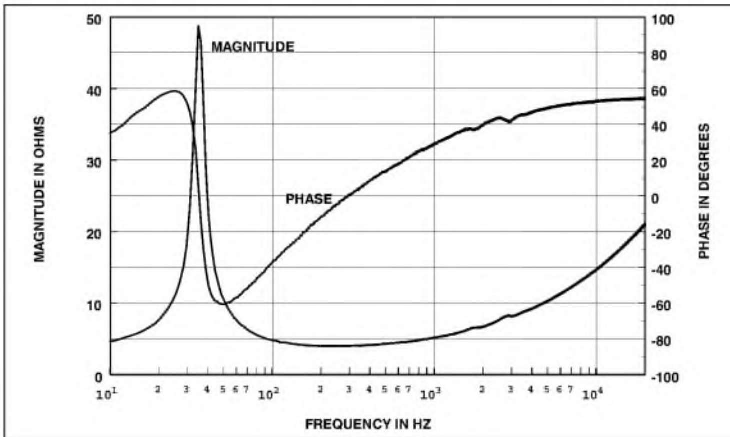



**FIGURE 3: Woofer impedance, 10Hz–20kHz.**

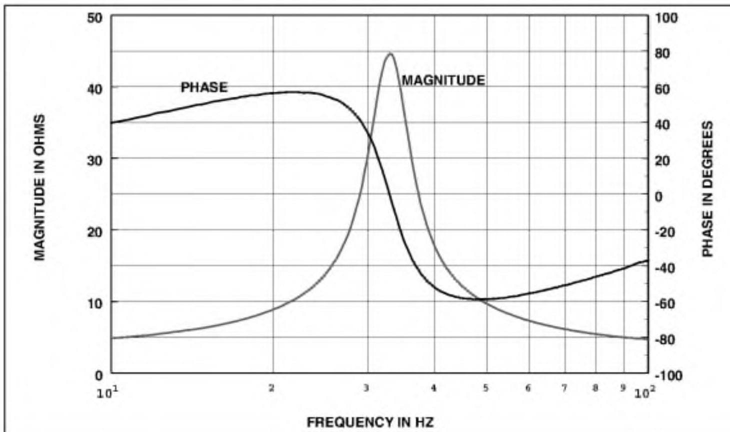



**FIGURE 4: Low-frequency detail.**

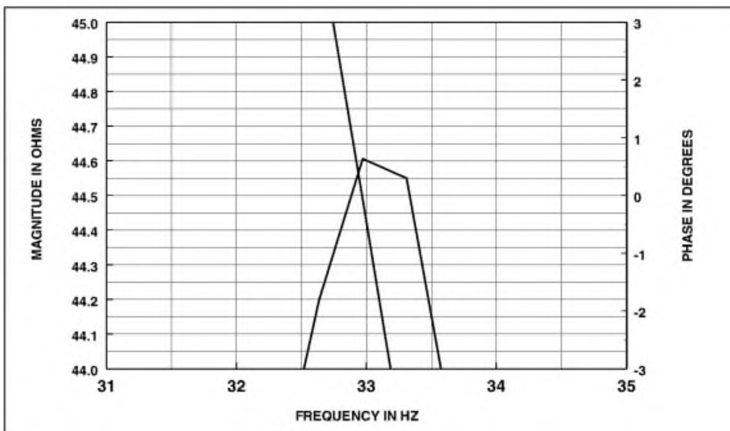



**FIGURE 5: Further low-frequency detail.**

## CRUCIAL SOUNDCARD SETTINGS

The soundcard I used in this study is a Creative Labs AWE64 Soundblaster, but any Soundblaster-compatible soundcard with full-duplex capability should work fine. Some of the soundcard settings are crucial for accurate results, and you can access these through the Windows volume controls. All balance controls should be centered (the default setting), and you should not adjust them unless they have been moved. If they have, you can most accurately center them by uninstalling and reinstalling the Windows volume control.

Set the main and wave volume controls to their maximum positions, and mute the line-in volume to prevent feedback. Set the recording line-in volume to its maximum position, and check its select box. Also, uncheck the recording microphone control select box to lower the system noise. With these settings, the maximum soundcard input and output levels correspond to 1V RMS, and the VU meters previously mentioned will accurately measure voltage in dB relative to 1V.

The impedance of the loudspeaker shown in *Fig. 2* is calculated from a transfer-function measurement computed by AtSpec. The transfer function, H(f), is given by

$$H(f) = SV_B(f)/SV_A(f), \qquad (1)$$

where $SV_A(f)$ and $SV_B(f)$ are the power spectra of voltages $V_A$ and $V_B$, and f is the frequency. The transfer function, also called the frequency-response function, is a complex quantity, as are the power spectra.

AtSpec computes the power spectra by performing fast Fourier transforms on the time signals $V_A$ and $V_B$. This is a conversion from the time to the frequency domain. Since the voltages are random noise, the analyzer must perform a number of averages to produce meaningful results. The transfer function is expressed by

$$H(f) = R + iI \qquad (2)$$

where R and I are the real and imaginary components of H(f), and $i = \sqrt{-1}$. H(f) can also be expressed in terms of its magnitude (M) and phase ($\phi$), given by:

$$M = (R^2 + I^2)^{1/2} \qquad (3)$$

$$\phi = \tan^{-1}(I/R) \qquad (4)$$

AtSpec computes both M and $\phi$, the quantities needed to obtain the impedance, which is obtained from the following equations:

$$Z_{MAG} = R_1 \{[M \cos\phi - 1]^2 + M^2 \sin^2\phi\}^{1/2} \quad (5)$$

$$Z_{PHA} = \tan^{-1}\{M \sin\phi/[M \cos\phi - 1]\} \quad (6)$$

The quantities M and $\phi$ are exported from AtSpec in ASCII text-file format, and the impedance is calculated and graphed using Axum, Excel, or Lotus 1-2-3.

## IMPEDANCE MEASUREMENTS

For example, I conducted impedance measurements on a Madisound model 6102-4 woofer both in free air and in sealed and vented enclosures. *Figure 3* shows the free-air impedance results from 10Hz to 20kHz using a 44.1kHz sample rate, 16,384 point FFT, and 200 averages. This results in a frequency resolution of 1.346Hz.

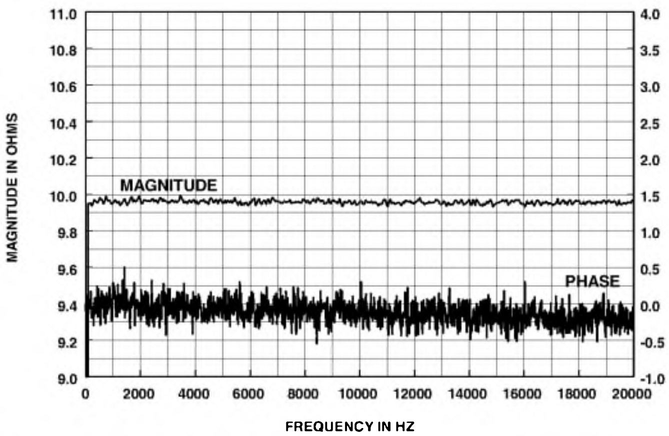



**FIGURE 6: Accuracy check.**

This wide frequency-range view is useful for crossover design, and you can determine the voice coil inductance from these curves. *Figure 4* shows a more detailed view of the impedance around the free-air resonance ($f_S$), measured with a sample rate of 11.025Hz and an FFT size of 16,384 points, and resulting in a frequency resolution of 0.3365Hz. A more detailed view of the *Fig. 4* data is shown in *Fig. 5*, where you can accurately read the magnitude of the impedance and $f_S$. The value of $f_S$ is the frequency at which the phase is zero, in this case about 32.9Hz, or about 10% higher than the manufacturer's published value of 30Hz.

I measured two other samples with similar results, varying about 5% and all higher than the 30Hz value, even after an extended break-in period. These are not excessive variations considering the manufacturing methods and low cost of most loudspeaker drivers, but they do affect loudspeaker system design.

As a check of the system accuracy, I substituted a 10Ω, 1%-precision resistor for the speaker under test in *Fig. 2*. The sample rate was 44.1kHz, with an 8192-point FFT and 200 averages. The results are shown in *Fig. 6*.

The magnitude accuracy is well within the 1% accuracy of $R_1$ and the test resistor; a 2% deviation would be quite acceptable, representing an audio measurement accuracy of 0.2dB. The phase accuracy of better than 2°, or 0.56%, is to be expected since the phase measurement does not depend on the soundcard-gain mismatch, but only on its phase mismatch.

You can measure the accuracy of the soundcard alone by repeating the described procedure with both line inputs connected to the same random-noise source. You can directly observe this on the AtSpec screen, and no exporting or calculations are required. The transfer-function magnitude should have a value of 1.0, and the phase should be close to zero. Any in-correct settings of the Windows volume controls will be obvious with this procedure.

## THE ANALYSIS OF .WAV FILES

All the examples here were done with real-time data acquisition of the analyzer from the soundcard. This works well, but there are some real advantages to recording the time-domain data on .wav files for later analysis. This is analogous to analog taping of data for permanent storage, which is particularly useful for analyzing low-frequency data. The analyzer can then perform any desired analysis on the same data with the .wav file without having to reacquire it from the soundcard, thus saving substantial computation time. You also have a permanent record of the raw data.

The resulting files are large. A one-minute .wav file recorded at a 44.1kHz sample rate produces a file of about 10.6MB. The same data recorded at 22.05kHz would be one half that size (5.3MB). The standard sample rates are 44.1, 22.05, 11.025, and 5.5125kHz. Some soundcards have additional choices. The Ensoniq cards, for example, will also sample at 4kHz. The usable bandwidth of the recorded data is one half the sample rate. A .wav file recorded at a rate of 44.1kHz will have an audio-frequency response that is flat up to 22.05kHz.

You can downsample the recorded .wav files to lower sample rates with software such as Cool Wave, resulting in lower bandwidths but increased frequency resolution for a given FFT size. Cool Wave will also low-pass-filter the downsampled data to prevent aliasing. The only advantages to downsampling are increasing the maximum frequency resolution of the analyzer and reducing the .wav file size.

For example, if you are interested only in the impedance of a woofer up to 100Hz, you can save a lot of disk space by downsampling the data file at a lower sample rate than the minimum soundcard rate of 5512Hz. The maximum frequency resolution of AtSpec for analyzing data from soundcard-recorded .wav files is 0.084Hz. Downsampling can increase this resolution, but you rarely need this increase in precision.

## IMPEDANCE MEASUREMENTS IN ENCLOSURES

*Figure 7* shows the impedance of the Madisound 6102-4 woofer in an 18.7 ltr sealed enclosure. You can combine the data of this figure and the free-air data of *Fig. 4* to obtain the Thiele/Small parameters for this woofer.[6] *Figure 8* shows the impedance of the same woofer in a vented 18.7 ltr enclosure tuned to 38Hz using the published Thiele/Small parameters. You can determine the actual box tuning, $F_B$, from *Figs. 7* and *8* with the following equation.[6]

$$F_B = (F_L{}^2 + F_H{}^2 - F_C{}^2), \qquad (7)$$

where $F_L$ and $F_H$ are the lower and higher peak frequencies of *Fig. 8*, and $F_C$ is the closed box resonant frequency from *Fig. 7*.

The design and measured-box tuning frequencies differ by about 10%, meaning that the design would be better using the measured, rather than the published, Thiele/Small parameters.

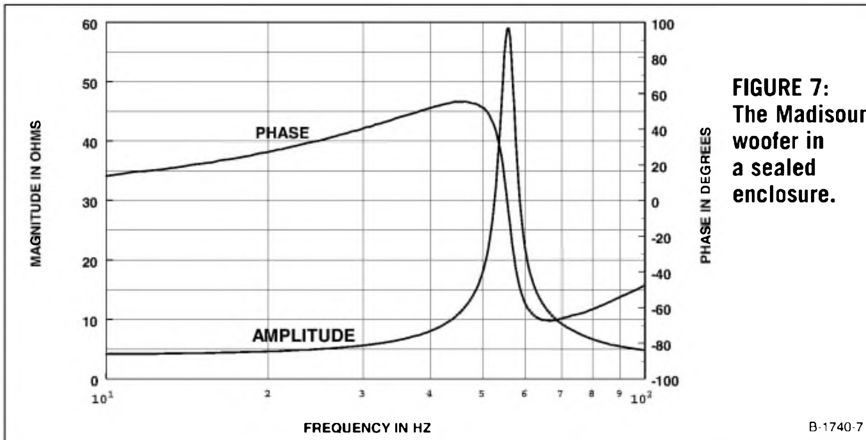



**FIGURE 7: The Madisound woofer in a sealed enclosure.**

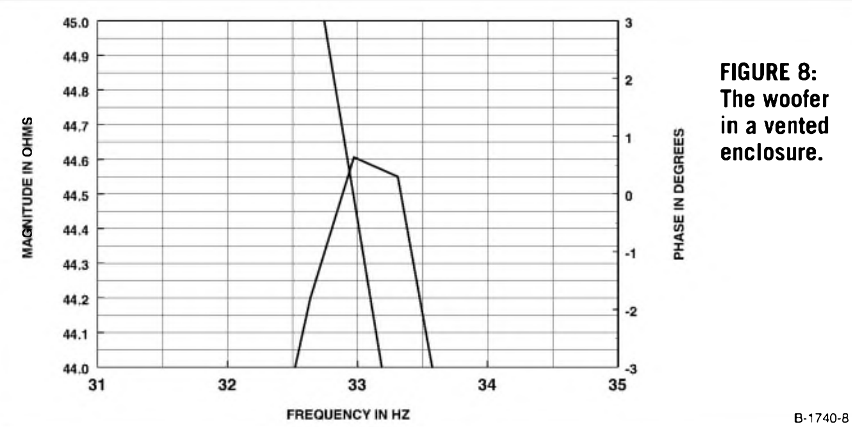



**FIGURE 8: The woofer in a vented enclosure.**

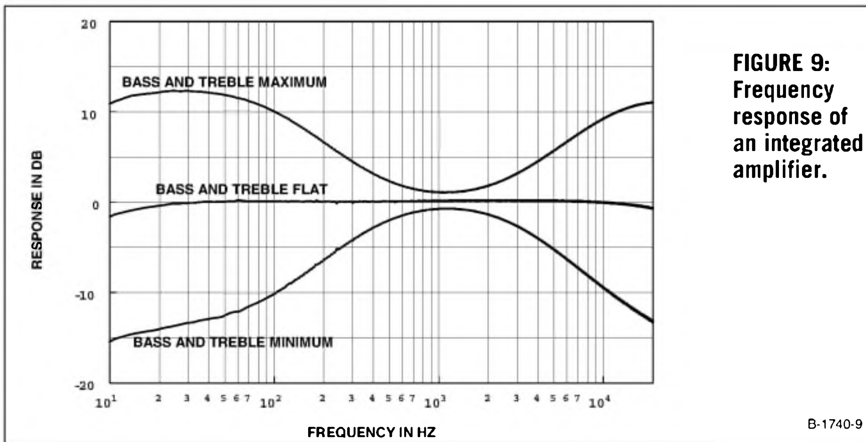



**FIGURE 9: Frequency response of an integrated amplifier.**

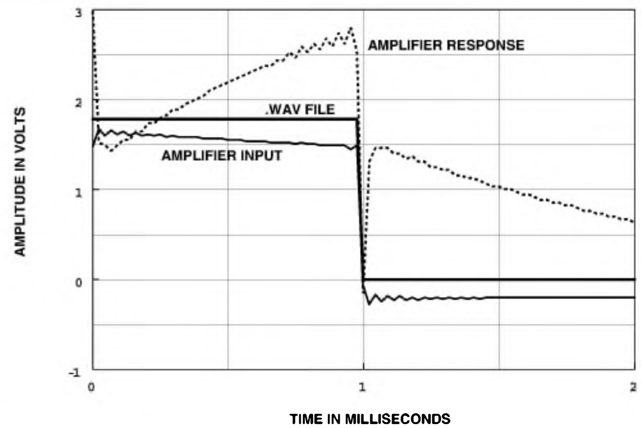



**FIGURE 10: Transient response of an amplifier.**

## OTHER APPLICATIONS

AtSpec and the other software described here convert the personal computer equipped with a soundcard into a group of audio instruments capable of generating, monitoring, and analyzing audio signals. With these, you can quickly and accurately measure frequency response and harmonic distortion of audio equipment. *Figure 9* shows the frequency response of an integrated preamp-power amplifier for various tone control settings. Steady-state and transient time domain signals are easily captured and stored in digital files.


### REFERENCES

1. Dick Pierce, "The Impedometer" *SB* 8/99, pp. 28–31.
2. AtSpec software by Paavo Jumppanen. See the author's web page at www.taquis.com.
3. Wavetools software by Paul Kellett. This software can be downloaded from the author's website at www.abel.co.uk/~maxim.
4. GoldWave software by Chris Craig. A fully functional demonstration version is available for downloading at www.goldwave.com. The full version, which eliminates a pop-up window at startup, is available for $40.
5. Cool Edit by Syntrillium Corp. A limited function demonstration version, which you can download at www.syntrillium.com, will do noise generation and downsampling described here and will record and play .wav files. The full version costs $50.
6. David B. Weems, *Great Sound Stereo Speaker Manual with Projects*, Chapter 10, TAB Books Division of McGraw-Hill, 1990.
7. Awave software by Markus Johnson (FMJ-Software). The free demonstration version, which will convert ASCII text to .wav files, can be downloaded from their website at www.fmjsoft.com.
8. The frequency-counter and sine-wave generator software can be downloaded free from the following website: www.espanet.com/caribu/software.html.
9. Test-tone generator software by Timo Esser. This freeware can be downloaded from the author's website at: www.esser.u-net.com.
10. Spectra Plus software by Pioneer Hill Software. A free fully functional (but time-limited) version can be downloaded at www.telebyte.com/pioneer.


*Figure 10* shows the time-domain transient response of the amplifier used in *Fig. 9* for a 1ms pulse input with both tone controls set to maximum. I used a simple BASIC program to generate the .wav file as an ASCII text file, and then converted it to a .wav file with Awave[7]. The resulting file is phase-inverted, but this you can correct with the invert function in GoldWave or Cool Edit.

I looped the .wav file with the Wavetools' signal generator and captured the waveforms with AtSpec in the one-shot trigger mode. That the amplifier input (which is also the soundcard line output) varies somewhat from the .wav file is due to the soundcard AC coupling and the very sharp cutoff of its antialiasing filters. The large deviation in the amplifier transient response is due to the tone-control settings. With these bypassed, the amplifier response is almost identical to the input.

Other useful and free audio-measurement software includes a frequency counter and sine-wave generator[8] and a test-tone generator[9]. The frequency counter will measure and display the frequency of an audio signal connected to the soundcard line input with an accuracy of better than 1Hz. The sine-wave generator has continuous manual settings of frequency and amplitude, and the test-tone generator produces swept and constant-frequency sine, square, and triangular waveforms.

## CONCLUSIONS

The software described here, along with a personal computer equipped with a soundcard and the Windows 95 or 98 operating system, allows you to conduct high-precision impedance measurements with exceptional frequency resolution. The data is exported in standard ASCII text format for further computation and plotting with readily available spreadsheet programs. The resulting graphs are of publication quality.

Many other audio measurements, such as waveform and level monitoring and transient and frequency response, are easily accomplished without the need of expensive test equipment. Installing the software on a laptop computer provides you with a very portable audio measurement laboratory.

The FFT analysis I describe here is also possible with another soundcard-based program called Spectra Plus,[10] which can also perform FFTs up to 32,768 points, but is considerably more expensive (over $600). It has a number of added features such as waterfall displays (three-dimensional time-frequency and amplitude analysis) and octave and one-third octave band analysis, and you may find it worth the additional cost if you need these additional capabilities. ❖

# Constant-Current Regulator

Here are some ways in which you can use a constant-current regulator and how you can build a low-cost version for experimental use. You can test in two ranges: 0–40mA and 0–400mA. **By John Stewart**

Ever wonder how you could make accurate measurements of small resistances, such as tube heaters or incandescent bulbs, whether hot or cold? How about semiconductor diodes? The results for all will depend on something you won't know: current applied by the meter. And what about audio or power transformer windings? An ordinary volt-ohm meter (VOM) or digital multimeter (DMM) does a poor job. Many experimenters have at least a passing experience with constant-voltage regulators and their application. This is probably not the case with constant-current (CC) regulators. With your meter and a constant-current source, you can make accurate four-terminal measurements, free from contact and lead resistance. You can also use it as a battery charger. Mine was built in 1969 and has been in use since. It looks and works well (*Photos 1* and *2*).

Yours does not need to look the same or even use the same parts. The design is simple and flexible.

## CIRCUIT DESCRIPTION

The Hammond 166G20 transformer, along with a diode bridge and 500µF cap, provides the required DC. A variable reference voltage is tapped off by R1, the 5k wirewound pot. The drop across the base-emitter junction of the 2N4143 is about 0.6V. If the test points are shorted, the resulting current depends only on the tapped-off reference voltage minus 0.6V applied to resistance in the 2N4143 emitter circuit (*Fig. 1*).

For example, if the tapped-off reference was set to 5V and S2 was in the low range (open), the resulting current would be (5.0–6.0)/220, or 20mA (4.4V across 220Ω). You can set the current by connecting a milliammeter to the test terminals. Once set, the current meter does not need to be left in the circuit. The regulator circuit will maintain this current up to the voltage limit of its supply, less a bit for drop across the transistors.

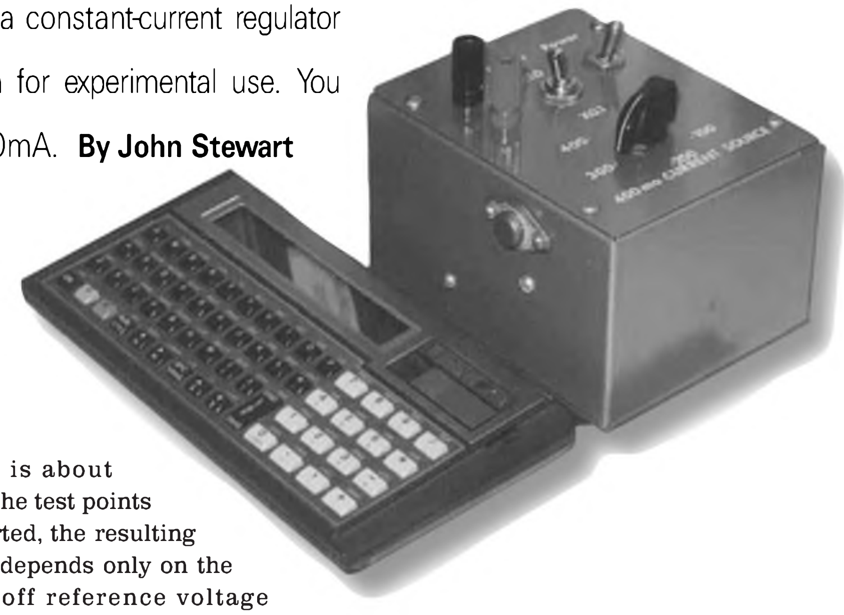


**PHOTO 1: Constant-current regulator.**

**TABLE 1**
**HAMMOND 271X WINDING RESISTANCE MEASUREMENTS APPLIED 20.1MA FROM CC REGULATOR**

| WINDING | MEASURED (MILLIVOLTS) | R CALCULATED (OHMS) |
|---|---|---|
| Primary | 135.9 | 6.76 |
| 6V | 4.2 | 0.209 |
| 5V | 3.6 | 0.179 |

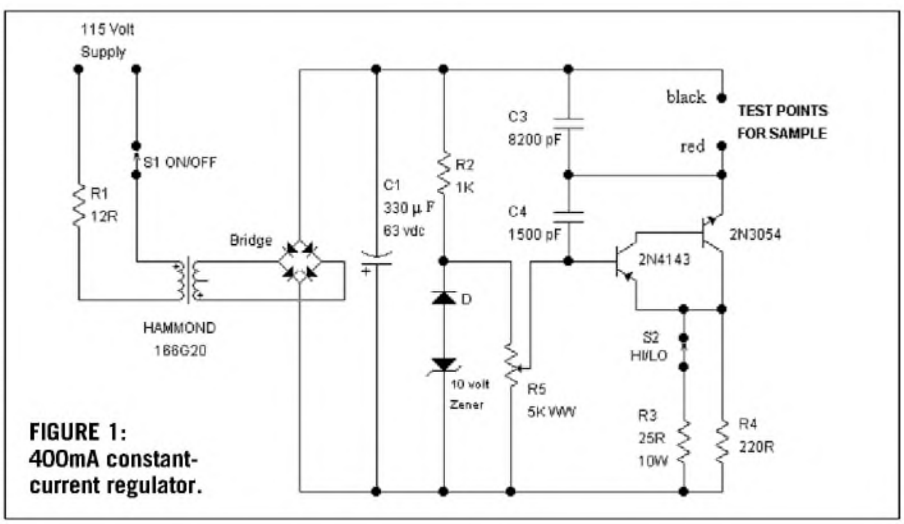



**FIGURE 1: 400mA constant-current regulator.**

**ABOUT THE AUTHOR**

John Stewart, a professional Engineer (Electrical) has finally retired after 35 years in the sales arena, primarily with Rohde & Schwartz and Hewlett-Packard. He gained design experience with vacuum tubes while working for the University of Toronto, Physics in the '50s and '60s. He now spends too much time on a mountain bike and worries about what odd jobs his wife will come up with.

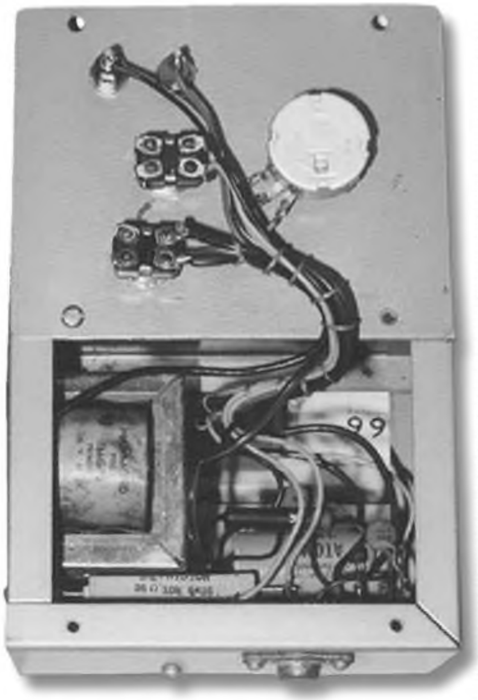


**PHOTO 2: Bottom view of regulator.**

If you now connect a transformer winding of 300Ω to the test terminals, you could measure 6V with the same VOM. To avoid contact-resistance-caused errors, the resulting voltage is measured on the transformer leads, not the CC regulator terminals (*Fig. 2*).

Closing S2 (the hi/lo switch) connects the 25Ω, 10W resistor in parallel with the 220Ω resistor. The parallel combination results in about 22.5Ω so that current output increases by 10×.

The silicon diode D in series with the zener compensates for temperature drift in the zener diode. I used a 12Ω, 1/8W resistor in the transformer primary to act as a fuse. It hasn't blown yet.

## CONSTRUCTION

I managed to get my version of the CC regulator into a 4″ × 3″ × 5″ box made by Hammond, but any box or chassis you find convenient will do. I used a terminal board made by Vector to mount all of the small parts. The terminal board approach works well if you intend to stuff everything into a small enclosure.

The 2N3054 power transistor is mounted on the outside surface of the box on its insulator. The box is made of steel, so you might think this would act as a poor heatsink for the 2N3054. That doesn't seem to be the case since the transistor never gets more than warm to the touch. There are no electrical connections to the box.

## APPLICATIONS

Twice in the past month I've found applications for my CC regulator. In one instance I wanted to make accurate measurements of power transformer primary and secondary resistance. I used a Radio Shack DVM to set up the conditions and measure the results (*Table 1*).

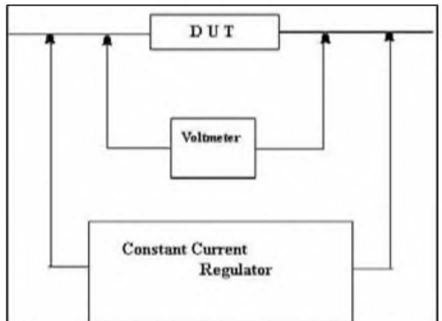



**FIGURE 2: Constant-current regulator test setup.**

**TABLE 2**
**PARTS LIST**

**400MA CONSTANT-CURRENT SUPPLY**

| | |
|---|---|
| C1 | 500µF 50V DC electrolytic |
| C2 | 1000pF 600V DC ceramic |
| C3 | 8200pF 600V DC ceramic |
| Hammond 166G20 transformer | |
| Hammond 1415B | 4 × 3 × 5 box or RS p/n 270-253 |
| S1, S2 | SPST switch RS p/n 275-651 |
| 10V | 2W zener |
| Binding post, black AE p/n S-H2131B | |
| Binding post, red AE p/n S-H2131R | |
| 2N3054 NPN transistor and mica insulating kit | |
| 2N4143 | PNP transistor |
| R1 | 12Ω, 1/8W |
| R2 | 1k, ½W |
| R3 | 25Ω, 10W |
| R4 | 220Ω, 2W |
| R5 | 5k linear pot |
| Bridge | 1 of AE p/n BR34 or 4 of AE p/n 1N4005 or 4 of AE p/n 1N4937 |
| D | 1N4005 or 1N4937 (any small power diode) |
| Knob | AE p/n P-K307 |
| Terminal board | |
| Feet, pkg. of 4, AE p/n P-H1308 | |
| Assorted screws, nuts, and washers | |

AE indicates Antique Electronics
RS indicates Radio Shack

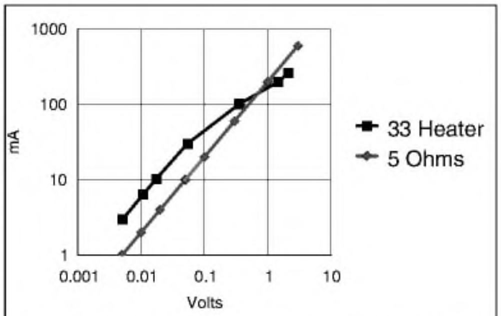



**FIGURE 3: 33 heater and 5Ω measurements.**

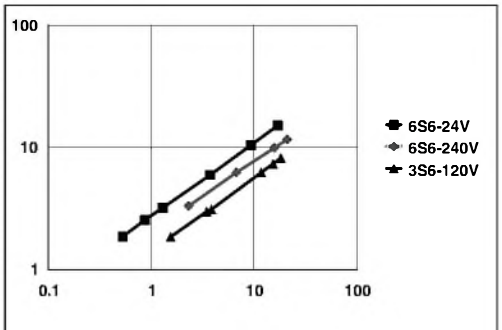



**FIGURE 4: Lamps comparison.**

The other case was when I needed to know the cold resistance of filaments in some power pentodes which are directly heated. Their resistance depends upon the applied current. When cold they measured 1.85Ω, but at the operating current they were 11.3Ω. For this kind of measurement you must wait while the filament temperature stabilizes. It takes one to five seconds. These kinds of measurements (*Fig. 3*) aren't possible with a VOM or DMM alone.

I have added the plot of a 5Ω resistor to the graph so you can see how an ordinary resistor compares to a tube heater. While the tube heater plot is above the 5Ω line, it is less than 5Ω. As the heater temperature rises, the resistance increases and the heater plot crosses the 5Ω line. When the heater plot is below the 5Ω line, it is more than 5Ω. This heater plot is typical of most metal conductors.

You might wonder how knowledge of electrical resistance of incandescent bulbs applies. One of the most famous electrical patents ever issued was to William Hewlett. It was for a Wien bridge audio oscillator, which used a small tungsten bulb as a stabilizing device in a feedback network. The HP200A oscillator became the first successful product to be marketed by Hewlett and David Packard.

## COMPARISONS

*Figure 4* is a comparison of three small lamps. The 3S6-120V was used by Heathkit in their version of the Bridged

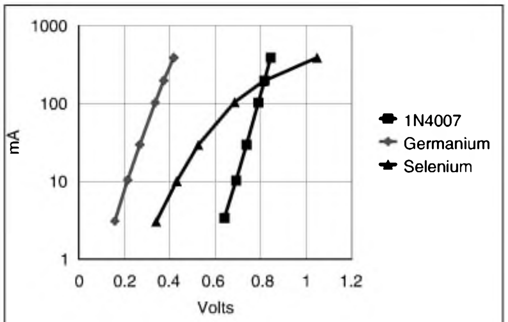


**FIGURE 5: Power diodes comparison.**

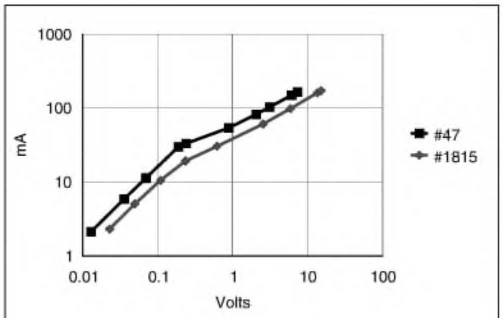


**FIGURE 7: Pilots comparison.**

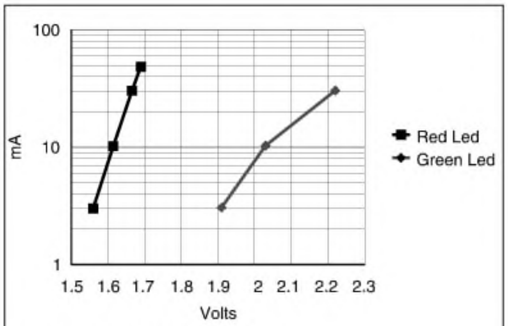


**FIGURE 6: LEDs comparison.**

**SOURCES**
www.hammondmfg.com
www.vectorelect.com
www.tubesandmore.com

"T" Oscillator. You would find two of the 6S6-240V in the Hewlett-Packard 200CD Wide Range Oscillator, which uses a Wien bridge. Measurement of these characteristics at each point could take up to 30 seconds while the lamp filament temperature reaches steady state after the test current is set.

*Figure 5* plots the forward characteristic of three common semiconductor diodes from different eras. You can see how the Selenium rectifier sags as the current increases. Nevertheless, they are still widely used in automotive battery chargers, since they are very forgiving of accidents. They are also used as surge suppressors because they usually have a large thermal capacity. Silicon diodes make very good low-voltage regulators, either alone or in series. Germanium power diodes are not used much anymore, but I still have some in my junk box.

*Figure 6* looks at conduction in light-emitting diodes (LEDs). You can see the large difference between red and green LEDs. Some designers use LEDs as low-voltage regulators, where zeners don't go. Unlike zeners, they are noise free. They are also good indicators that the circuit is or isn't working.

*Figure 7* shows the characteristics of two common incandescent pilot lamps.

I have used logarithmic scaling for these measurements (*Figs. 3-7*) since that allows presentation of the results over a wide range of currents. ❖

# Odds and Evens

Sometimes two wrongs *can* make a right, as this designer from Australia demonstrates with his design that uses an even number of audio stages to improve amplifier linearity. **By Graham Dicker**

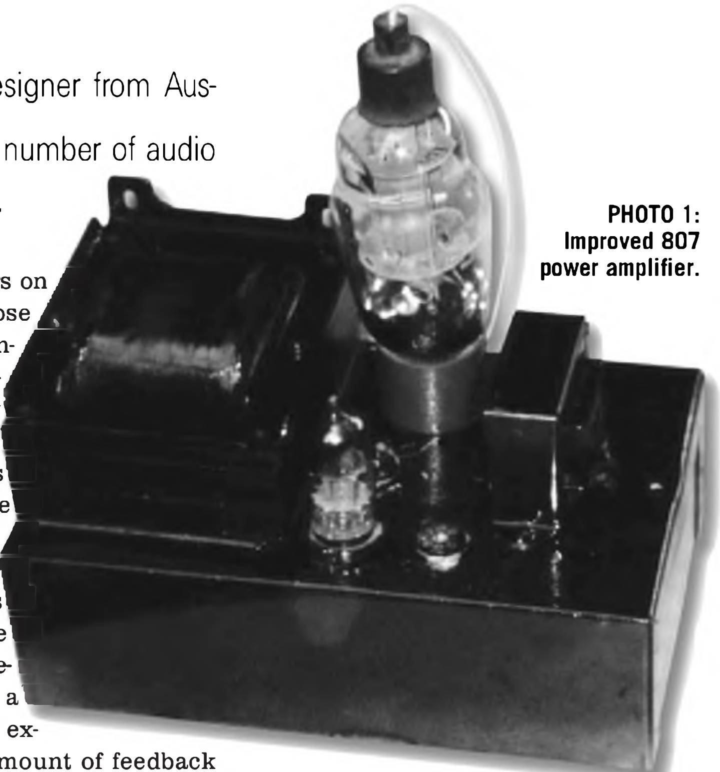


**PHOTO 1: Improved 807 power amplifier.**

Despite the rather cryptic title of this article, the underlying investigations have substantial ramifications in the area of high-end audio design. The principle is so simple that I wonder why no one has thought of it before.

## THE NONLINEAR PROBLEM

All amplifiers use active devices to produce current, voltage, or power gain. The problem with all such devices is that they are nonlinear; i.e., they have a transfer function whereby, for a given input, the output will be greater in one or more of the above respects, but not an exact version of the input. This is where the holy grail of audio comes in—trying to make the output as close to the original as possible. The standard technique is to use feedback of varying degrees and in different places to linearize the amplifier (even some commercial amplifiers that claim to have zero feedback often have internal local feedback).

The top amplifiers on the market are those that offer the best linearity and performance before any feedback is applied. These products often use expensive linear-transmitter output tubes to help achieve this (the triode is still the most linear active device available). And a marked trend now exists to make the amount of feedback used in an amplifier user-controlled.

My discovery is that distortion in any one of two voltage-amplifier stages is canceled by the second by the nature of the two transfer curves. The output phase of a single stage of a voltage amplifier is reversed from its input, so with two identical stages, any nonlinearity in one stage will be canceled by an equal and opposite nonlinearity in the second. If you add a third stage, you are back to the equivalent of a single transfer and its high distortions. Hence two wrongs (nonlinearities) make a right (linear) stage. The key here is that all voltage-amplifier stages must have even numbers of active devices to cancel out the distortion.

If you are alert, you will have realized by now the weak link in my argument—the presupposition that the two voltage amplifiers must have identical loads, characteristic curves, and operating conditions. This is so if you aim at an extremely low distortion figure, but it is not impossible to use a power-triode output tube as a voltage amplifier if you wish.

In practice, because all tubes have the same basic transfer curve, although differing degrees of sag, the principle still holds. The THD level from a 300B power and voltage amplifier is around .5%, in contrast to the 1.2% from a 300B output device and a 12AX7 voltage amplifier. With three voltage-amplifier

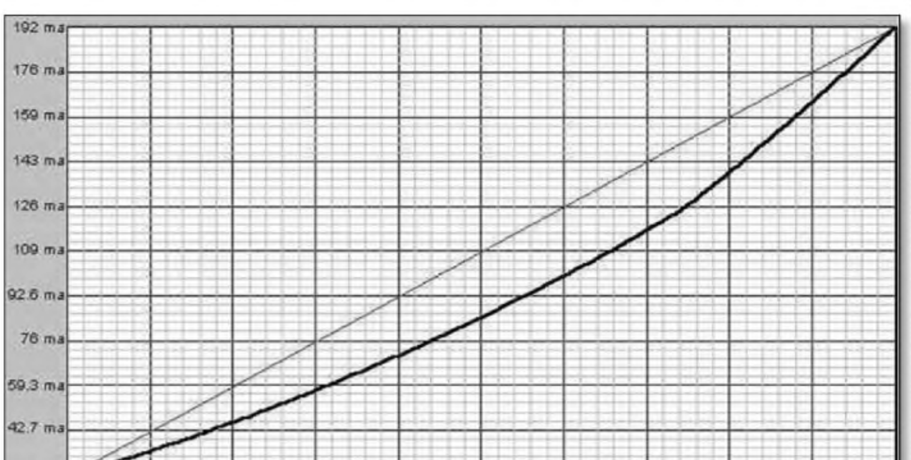



**FIGURE 1: Transfer curve of an 807 tube.**

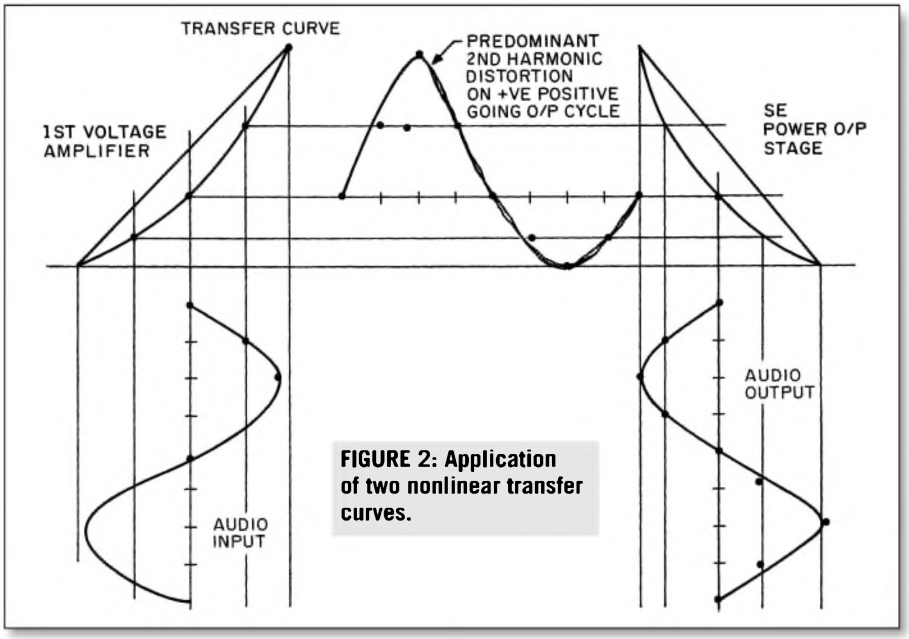



**FIGURE 2: Application of two nonlinear transfer curves.**

stages giving the same voltage gain, this figure rises to about 16%, more than ten times as much.

## OTHER INTERESTING ISSUES

Have you ever wondered, when you look at music on an oscilloscope, why the waveform was not symmetrical about the x-axis, or why reversing the phase of both speakers with some music improves the sonic qualities, or why guitar amplifiers sound terrible when reproducing music, and hi-fi amps sound terrible for live music. Furthermore, why does the 300B sound better than a lot of other tubes, and why do some single-ended (SE) amplifiers sound much better than others for no apparent reason. For that matter, why do tube amplifiers sound better than anything else?

Detailed investigation into amplifier design has shown that an even number of amplifier stages in any design, be it solid-state or tube, has substantial advantages. The trend today is toward designs with lower or no negative feedback at all. To achieve reasonable performance, the key issue is stage linearity. Many manufacturers bend these principles by incorporating local negative feedback within a stage or hiding it elsewhere in a circuit.

Regarding feedback, I believe that if the open-loop gain of a power amplifier having no negative feedback is less than 1%, then small amounts (less than 6–10dB) of negative feedback can often sonically improve its performance.

My investigations have involved examining the performance of a number of respected amplifier designs, using a detailed analysis of the circuit through computer modeling, and then sonically and electronically testing the performance of these units with all feedback removed.

## TRANSFER CURVES

The secret to understanding this even-number principle lies in active-device transfer characteristics. If you plot these for all known active devices, you discover that the most linear is the triode tube. Of the power output devices, the most linear are large transmitting tubes, followed by devices like the 2A3, 300B, and, surprisingly, the only two power tetrodes ever designed for use as triodes—the 6CA7 (EL34) and the 807. The key here is that the device's transfer "curve" is not a straight line, but actually a curve (*Fig. 1*).

If you use a sine-wave input to extract the output from this transfer function, the result is more gain on positive-going half cycles (*Fig. 2*). This results in second-harmonic distortion, where as the amount of nonlinearity of the curve directly increases, so does the second-harmonic distortion. Now if you are a musician and this is a guitar

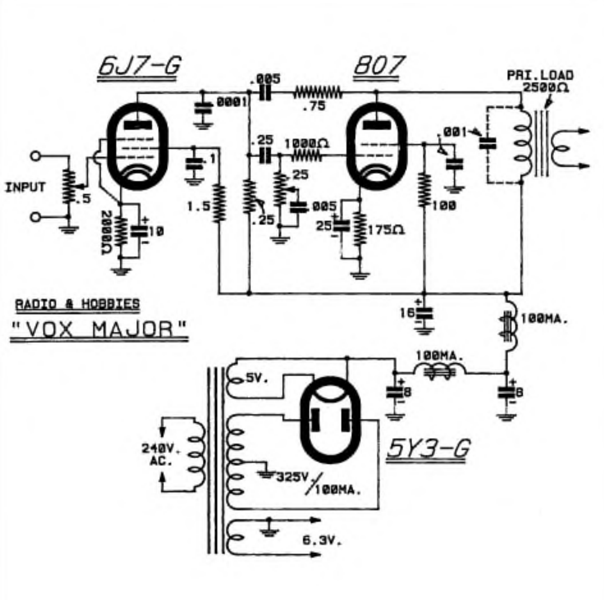



**FIGURE 3: Circuit of 1947 VOX MAJOR.**

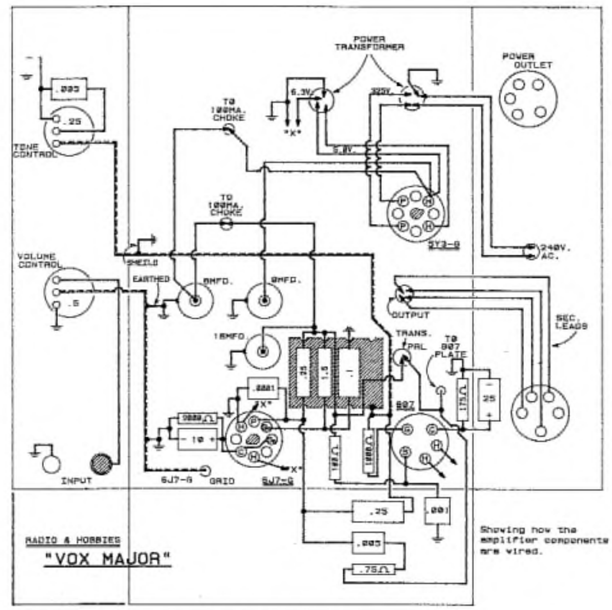



**FIGURE 4: Parts layout of VOX MAJOR.**

amp, the resulting overtones are quite warm and pleasant, but if it's a hi-fi amp, most people would not be at all happy.

If you apply the same transfer curve a second time, the first cancels out, leaving an undistorted output, which is also supported by testing. This reduces the overall second-harmonic distortion substantially. Even more interesting, if you now add a third transfer, you are back to where you started, with lots of second-harmonic distortion, so you can make the cycle go around. The point here is that an even number of gain stages reduces distortion, and an odd number increases it.

## CAD EVALUATION

One of the most outstanding designs I've encountered over the years is one I built 35 years ago from an original design in the April 1947 *Radio and Hobbies* magazine, using a single-ended 807 VOX MAJOR (*Fig. 3* with a copy of the circuit included; redrawn in *Fig. 4*). After a little resurrection work, this amplifier sounded better without feedback than most top-end amplifiers I tested,

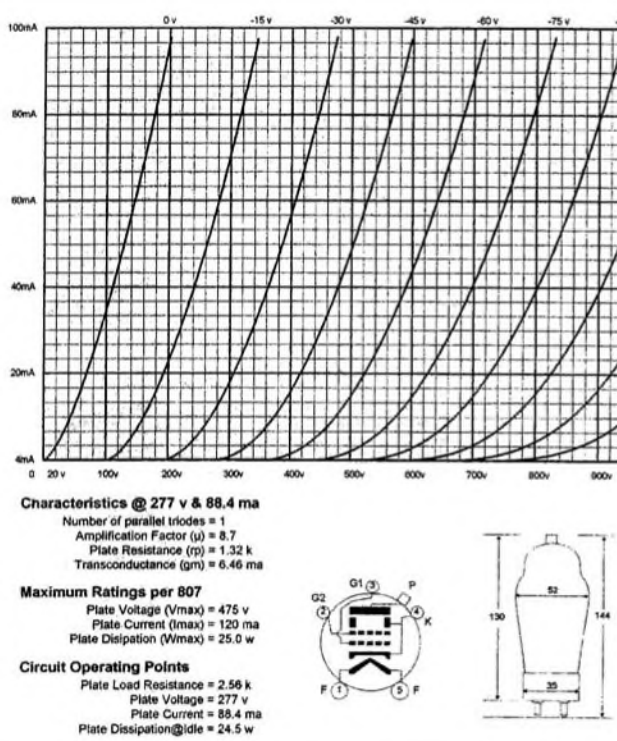



**FIGURE 5: 807 plate characteristics.**

so a little investigation was warranted, with the result that I used this unit as a test bed for my research work.

Two of my most useful tools are software products from GlassWare (PO Box 2, Santa Cruz, CA 95063, 408-438-5778, salessvc@tubecad.com), Audio Gadgets and SE Amp CAD (about $99 Australian each). These are great for modeling tube stages. My first job was to identify the Q point for the stage from the 807 plate curves (*Fig. 5*) and plot the load line of the original design (*Fig. 6*). As you can see, when this work was done over 50 years ago, the designer got it right—it is within the maximum current, voltage, and dissipation limits. The question was whether I could improve on the performance with the hindsight of 50 years of others' experience in amplifier design.

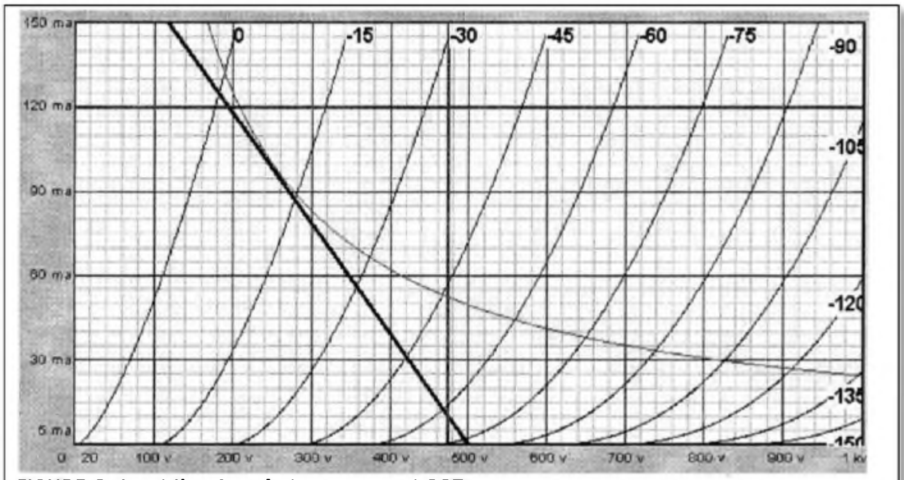


**FIGURE 6: Load line for triode-connected 807 output stage.**

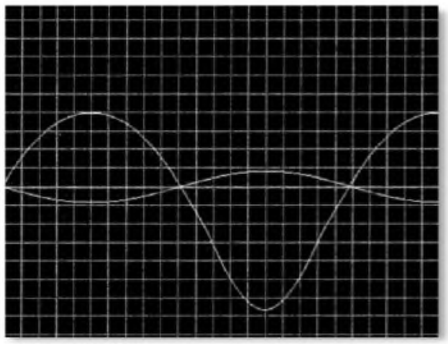


**FIGURE 7: 807 sine-wave input and output test.**

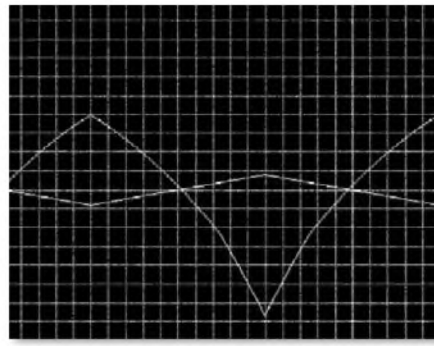


**FIGURE 8: 807 triangle-wave input and output test.**

## TESTING

In SE-amplifier testing, sine-wave or computer-based tests take a lot of time to reveal what's happening. A simpler and better method is to use a triangle waveform to test where the nonlinearities are. With the sinewave test (*Fig. 7*), the distortion is not easy to see, whereas in *Fig. 8* the curve on normally linear sections of the waveform quickly shows the stage nonlinearity. This has been the main method used for some time to evaluate amplifier and individual stage performance and linearity.

## IMPROVEMENTS TO THE 1947 DESIGN

My first step in improving the 1947 design was to convert to a triode-connected 807 by tying the screen to the anode with an 82Ω, 1W resistor, and then to calculate the performance of the triode-connected stage. An 807 can also be triode-connected in zero-bias mode by connecting the screen grid to the control grid. This unfortunately requires you to drive the device into grid current, which needs a very low output impedance in the preceding driver stage to keep transient-intermodulation distortion low, and is therefore not the method I used here.

The analysis of the output-stage data is shown in *Fig. 9* along with a slightly changed Q point, where the cathode re-

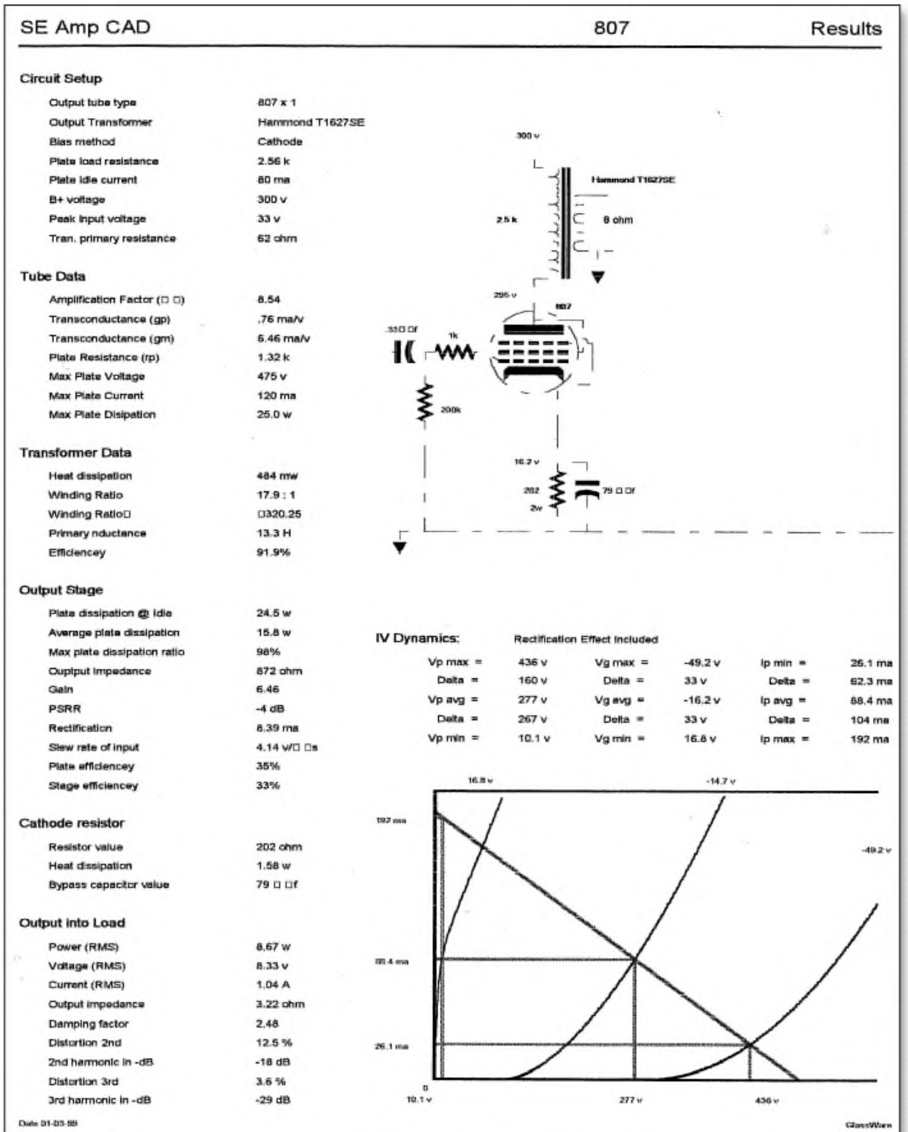



**FIGURE 9: Design parameters for 807 output stage (SE Amp CAD).**

sistor is now 202Ω instead of 175Ω. I removed the tone control and replaced it with a 220k resistor. I also entirely removed the negative feedback from the 807 anode to grid, and increased the grid-coupling capacitor in value from .25µF to .33µF to provide a lower-corner turnover frequency of about 2Hz. (To be phase-linear at 20Hz, the lower-corner frequency needs to be one-tenth of this value.)

In any amplifier chain there should be only one dominant pole, and in this case it is determined by the output-transformer primary inductance of about 14H. All preceding stages should have the same lower-corner frequency. In many pre-1970 designs that did not follow this principle, it was a cause of many amplifier stability and performance ills that could have been avoided (and this is a simple area in which you can improve the performance of many existing amplifiers).

The input stage originally used a 6J7 pentode, which was noisy, to say the least. I replaced this by half of a 12A3 of military manufacture, with specs similar to a 2A3, but a dual-triode device in a 9-pin base. Using a 2A3 or 6A3 would be equivalent. The new design parameters for the voltage amplifier are shown in *Fig. 10.* I removed the volume control entirely (it was noisy due to the grid space charge, causing DC to flow through the pot) and replaced it with a 1M resistor.

Another major change I made was to remove all electrolytics from the cathodes of the stages. A problem here is that simply removing them increases the effective anode resistance and also the output impedance, neither of which is desirable. A tidier way to solve the problem is with fixed bias, but the simplest solution is an LED in the cathode of the first voltage amplifi-

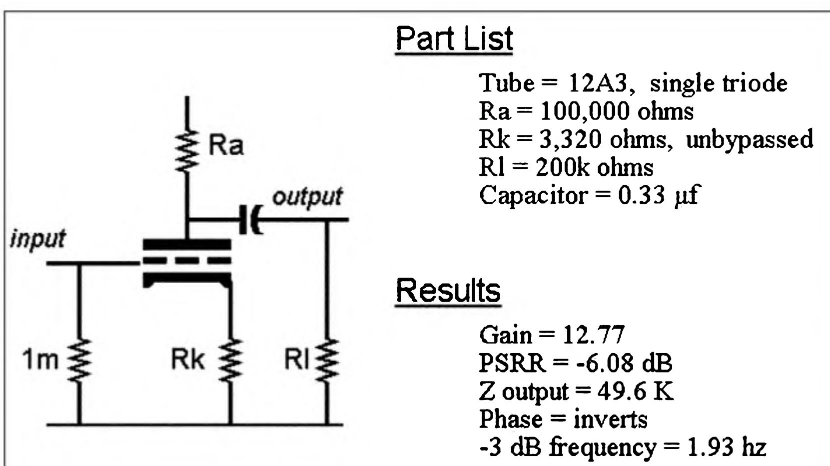



**FIGURE 10: Voltage-amplifier design (Audio Gadgets).**

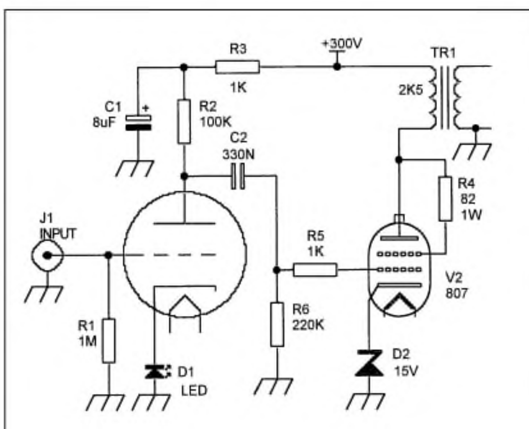



**FIGURE 11: Improved 807 power amplfifier V1.0 schematic.**

er and a 15V zener (1W) in the cathode of the 807. These also help keep things stable as the tubes age (*Fig. 11*). *Table 1* is the parts list.

The output transformer I used in the rebuilt design is a Hammond T1627SE (*Fig. 12*), available from Antique Electronic Supply (6221 S. Maple Ave., Tempe, AZ 85283, 602-820-5411, FAX

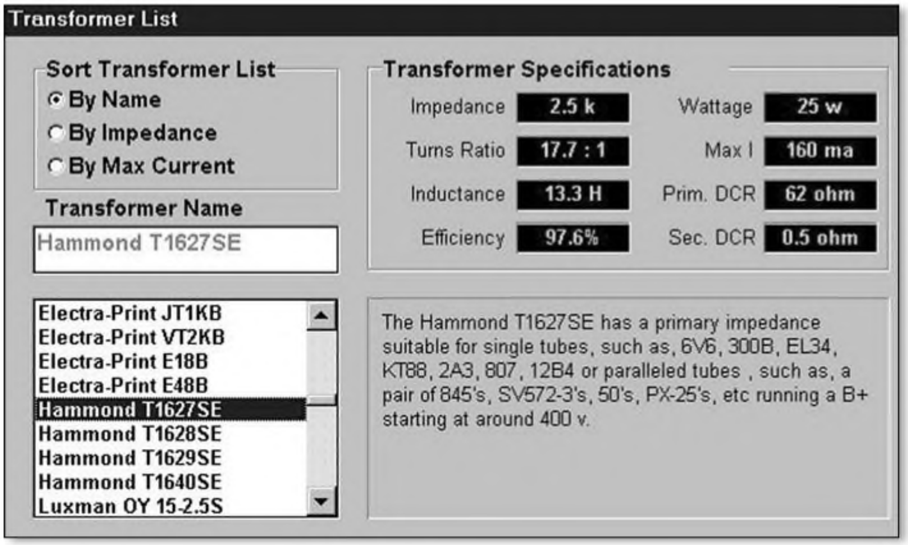
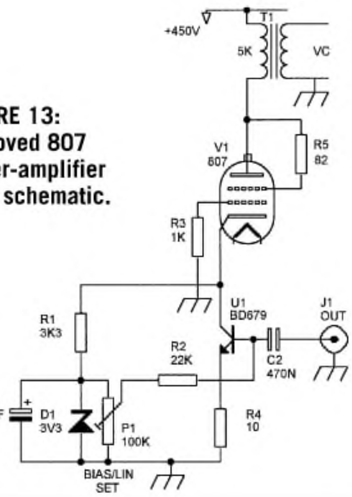



**FIGURE 12: Hammond T1627SE transformer specifications.**

**FIGURE 13: Improved 807 power-amplifier V2.0 schematic.**




| **TABLE 1**<br>**PARTS LIST: IMPROVED TUBE AMPLIFIER** |
|---|
| 1 × 100k ¼W metal-film resistor |
| 1 × 1M ¼W metal-film resistor |
| 2 × 2k ¼W metal-film resistor |
| 1 × 220k ¼W metal-film resistor |
| 1 × 1k ¼W metal-film resistor |
| 1 × 82Ω 1W metal-film resistor |
| 1 × 330nF (.33µF) polycarbonate capacitor (Philips) |
| 1 × 8µF 450VW electrolytic capacitor (Elna) |
| 1 × Red LED (HP) |
| 1 × 15V zener 3W (or 2 × 1W in parallel) |
| 1 × 807 tube |
| 1 × 12A3 dual triode (12AT7 as an alternative) |
| 1 × Hammond T1627SE output transformer (Antique Electronic Supply) |

| **TABLE 2**<br>**PARTS LIST: IMPROVED TUBE AMPLIFIER V2.0** |
|---|
| 1 × 100k "A" linear potentiometer (P1) |
| 1 × 10Ω ¼W metal-film resistor |
| 1 × 22k ¼W metal-film resistor |
| 1 × 3k3 ¼W metal-film resistor |
| 1 × 1k ¼W metal-film resistor |
| 1 × 82Ω 1W metal-film resistor |
| 1 × 470nF (.47µF) polycarbonate capacitor (Philips) |
| 1 × 100µF 10VW electrolytic capacitor (Elna) |
| 1 × BD679 npn Darlington transistor (Philips) |
| 1 × 3.3V zener (250MW) |
| 1 × 807 tube |
| 1 × Hammond T1627SE output transformer (Antique Electronic Supply) |

| **TABLE 3**<br>**PARTS LIST: TUBE AMPLIFIER SIMULATOR** |
|---|
| 1 × LM358 op amp (National Semiconductor or TI) |
| 1 × npn transistor |
| 1 × 1N914 diode |
| 1 × 2M2 ¼W metal-film resistor |
| 3 × 1M ¼W metal-film resistor |
| 4 × 10k ¼W metal-film resistor |
| 1 × 47k ¼W metal-film resistor |
| 1 × 1k ¼W metal-film resistor |
| 1 × 22k "A" linear potentiometer (P1) |
| 1 × 22k "C" log potentiometer (P2) |
| 6 × 100nF (.1µF) polycarbonate capacitors (Philips) |
| 2 × 22µF solid tantalum capacitors (tubular KEMET) |
| 2 × 330µF electrolytic capacitors (video-grade Rubicon—Japan) |



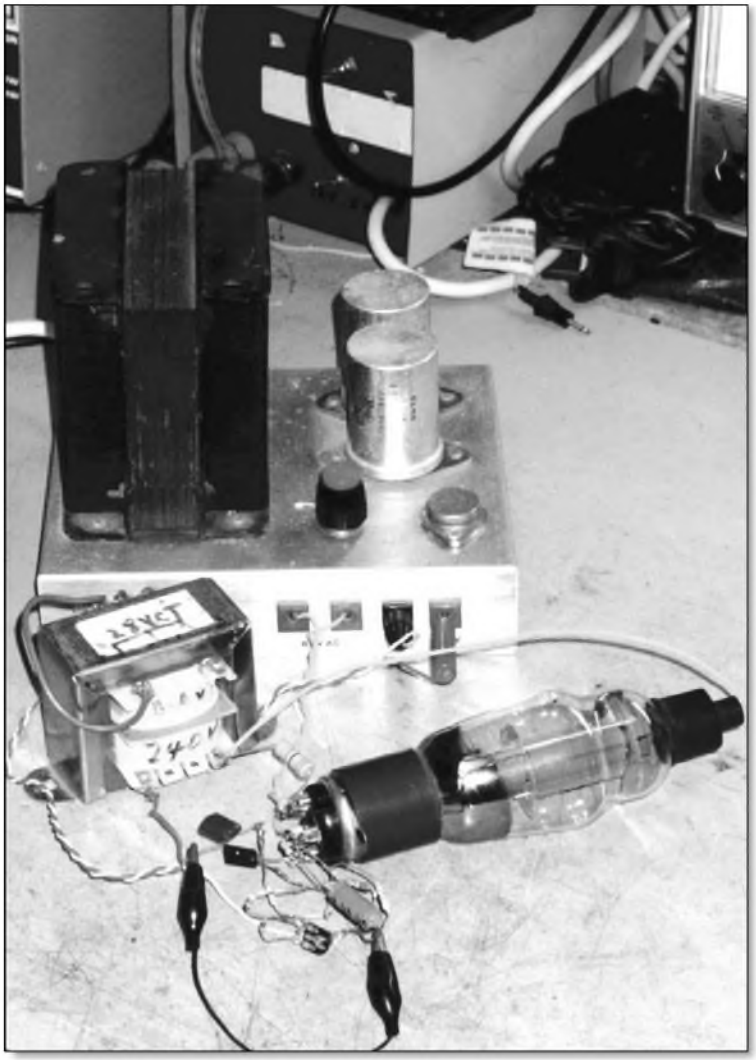


PHOTO 2: My test setup, with power supply, output transformer, and circuitry.

PHOTO 3: Closeup of "Christmas tree" construction (see Fig. 13).

PHOTO 4: Prototype preamp chassis.

602-820-4643) for about $250.

## SOMETHING A LITTLE DIFFERENT

While playing with different ideas, I decided to try a constant-current source in the cathode of the 807 to improve the linearity of the stage. This worked reasonably well, and then I had the idea of building a grounded-grid output stage (*Fig. 13*), thus removing the Miller effect. The parts list is in *Table 2*. The conversion was simple, and what was originally a constant-current source was now modulated with audio having an overall voltage gain of about 300, and a performance almost as good as the version in *Fig. 11*.

I moved the Q point for a higher plate voltage, replacing the original 5Y3 rectifier with silicon diodes so that the HT was 450V instead of 300V, and the plate load increased to 5k.

The circuit certainly has merit over many other designs sonically, but the prospect of a bipolar gain stage, silicon rectifiers, and a tube power amplifier is too much for many purists. The revised amplifier is shown in *Photo 1*.

## A TUBE AMPLIFIER SIMULATOR

Part of my investigations led me to analyze what really makes a tube amplifier sound the way it does, and to play with a little gadget I call the tube amp simulator (*Fig. 14*). This comprises an

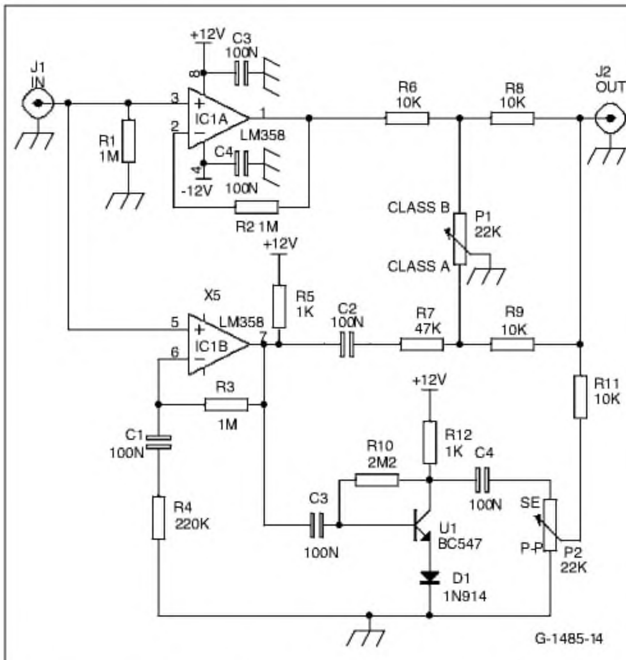



FIGURE 14: Tube amplifier simulator V1.0 schematic.

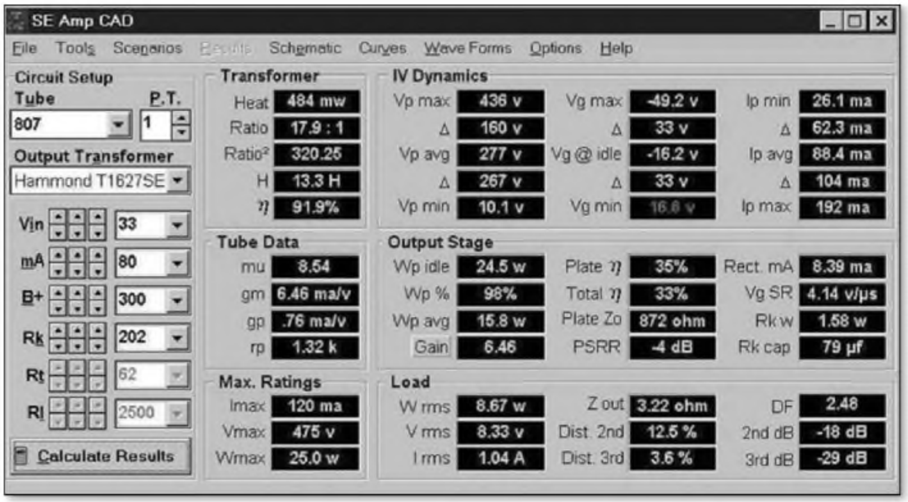



**FIGURE 15: Prototyping experimental tube circuits with SE Amp CAD.**

LM358 op amp, which, without an external current pullup resistor, exhibits crossover distortion. This makes a great Class-B simulator. The BC547 gain stage and diode in the emitter provides transfer characteristics similar to an SE triode stage by using the two pots. You can select lots of second-harmonic distortion (SE mode) or none (push-pull), Class A right through to Class B. This little gadget can add warmth to many a solid-state amp. If you can't afford a tube design, this may be the very device you have waited for. *Table 3* is the parts list.

It may seem strange to use so many bypass capacitors, but for good broadband performance, I used three separate ones. The main electrolytics are of video grade, offering low leakage and good HF performance. The 22µF devices cover good performance above 10kHz, with the 100nF devices covering from 40kHz upwards. Because of its simplicity, I constructed the prototype on a 2″ × 2″ section of vero board.

You may be interested to see how I prototype experimental tube circuits (*Fig. 15* and *Photo 2*). You can see my "Christmas tree" version of the circuit (*Fig. 13* and *Photo 3*), which works perfectly apart from some induced hum and noise. *Photo 4* shows my prototype preamp chassis, which simply contains two 9-pin valve sockets. All wiring is simply point-to-point, but everything is accessible. The amazing thing with tubes is how stable they are, regardless of construction. ❖

**SOURCES**

**Antique Electronic Supply**
6221 S. Maple Ave.
Tempe, AZ 85283
602-820-5411
FAX 602-820-4643

# The Virtual Crossover, Part 3

In Part 3, the author shows how you can use his Virtual Crossover to design and implement a crossover for a Roger Sanders-type hybrid ESL/TL speaker. **By Richard Mains**

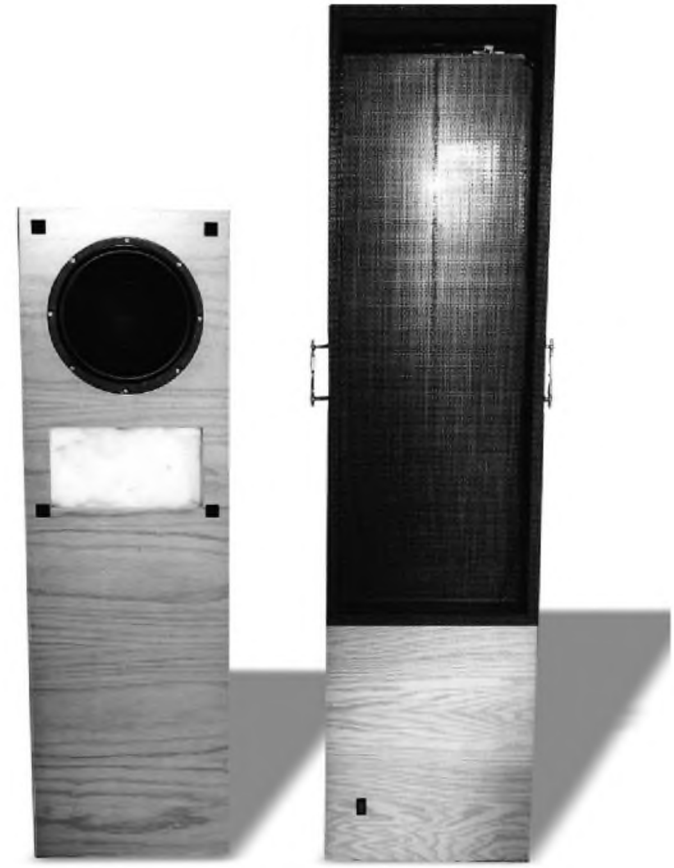


**PHOTO 4: One channel of the hybrid ESL/TL speaker system.**

I have previously made active crossovers for the Roger Sanders system using op-amp circuits with third-order Butterworth crossover filters and low-frequency ESL compensation similar to that which Sanders describes in his articles[1]. While this approach can yield very flat amplitude response for this system, the phase shift introduced by the third-order filters causes distortion of transient waveforms.

For example, in the square-pulse test with an ideal pulse waveform applied at the input of the speaker, it is not possible to obtain anything that resembles a square pulse at the output. However, using digital filters to simulate the crossover circuits, it is easy to remove the phase shift introduced by the third-order filters, resulting in a much improved square-pulse response for this system.

Whenever possible, I prefer to design speakers that exhibit good square pulse response, for at least two reasons. First, I have found that the speakers I like best over the long term are those with outstanding pulse response. Also, I regard the square pulse response as an important design tool; knowing exactly what a square pulse should look like, it is easy to see qualitatively any distortion in the speaker output.

Since I designed the Virtual Crossover to act as a measurement device as well as to simulate crossover circuits, I was able to use it to obtain the measurement results I present in this article. To carry out the measurements, I used filtered pulses that cover various frequency ranges, as well as FFT analysis. I still required a separate microphone preamp for the measurements because I used a microphone that requires phantom power, a feature that is not incorporated in the Virtual Crossover.

I used the "quasi-anechoic" measurement technique, which largely eliminates room effects without using an anechoic chamber. I accomplished this by truncating the speaker pulse response before it becomes dominated by room interactions.

## DRIVER EQUALIZATION

My first step in designing the crossover was to look at the individual driver responses without any crossover in order to determine what I needed to do to make each driver reasonably flat in the frequency range where it would be used. This is a two-way system, with a transmission line on the low end and a 13″ × 37″ electrostatic panel on the high end, and the crossover frequency between the two drivers is 500Hz.

Each driver must have a separate power amplifier to use the Virtual Crossover, so this speaker requires two power amplifiers. The ESLs are driven through step-up audio transformers. *Photo 4* shows one channel of this system. A step-up transformer and a high-voltage DC power supply are mounted out of view in the base of the ESL cabinet.

For the transmission line, I used the W1038R 10″ driver, available from Meniscus[2]. *Figure 19* shows a close-mike measurement of the response of this driver mounted in a transmission line (the microphone, a Brüel & Kjaer 4006, was placed right at the outer edge of the driver, so that room interactions were negligible). You can see that above 350Hz, the output of this driver rises to a peak of about 4dB at 884Hz, and there is a second peak at 1.641kHz. I decided that circuits should be included in the TL crossover to bring these peaks down to the level at 200Hz.

The first stage in the upper circuit of *Fig. 20* shows how I accomplished this. I placed two series-resonant circuits in the feedback path of op amp A1, each tuned to a peak in the woofer response. At that frequency, the resistor in the series path is effectively placed in parallel with R2, thereby lowering the response in the vicinity of the series-resonant frequency. I chose the resistors in the series-resonant paths, R3 and R4, essentially by trial and error until I was able to bring the woofer response at the peaks down to the low-frequency level.

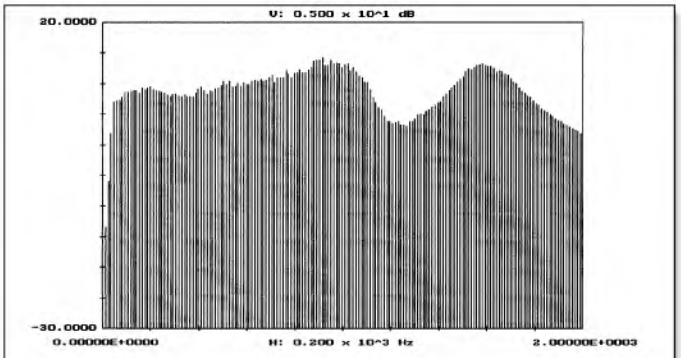



**FIGURE 19: Close mike measurement of Meniscus W1038R 10″ driver mounted in transmission line.**

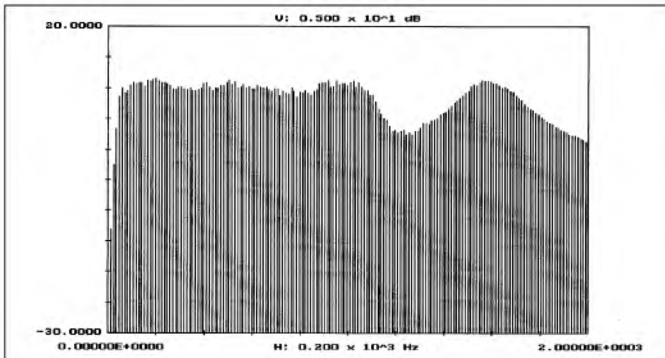



**FIGURE 21: Close mike measurement of Meniscus W1038R 10″ driver including the woofer trap circuit of *Fig. 20.***

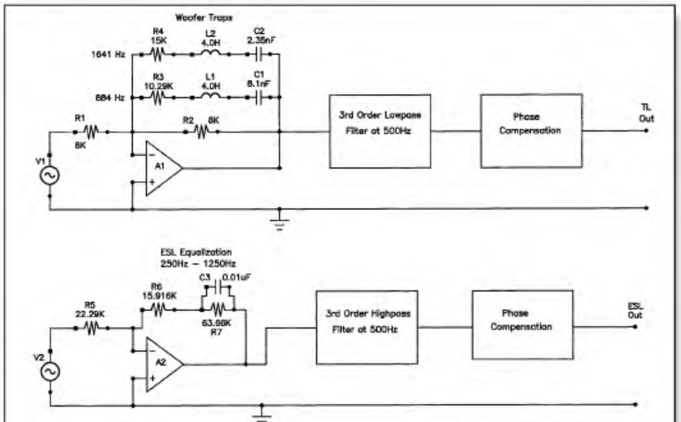



**FIGURE 20: Driver equalization circuits used for the TL (top) and ESL (bottom) crossovers.**

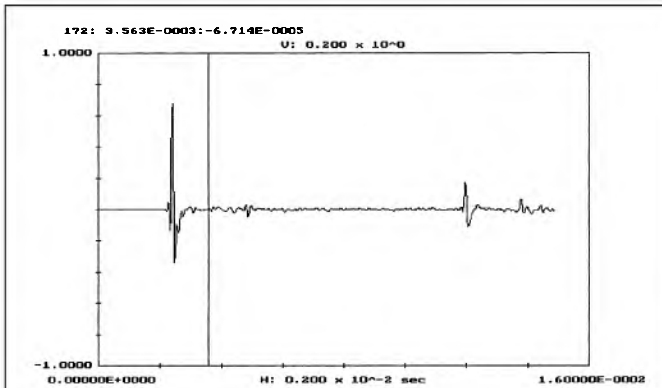



**FIGURE 22: Measured ESL pulse response to filtered pulse designed for measurements out to 5kHz. The response is truncated after the vertical line at about 3.5ms.**

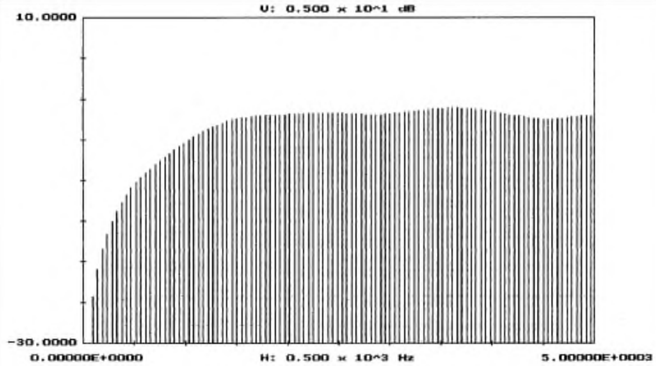



FIGURE 23: Amplitude response of ESL panel out to 5kHz with microphone about 6 ft. away, based on the measured data of *Fig. 22.*

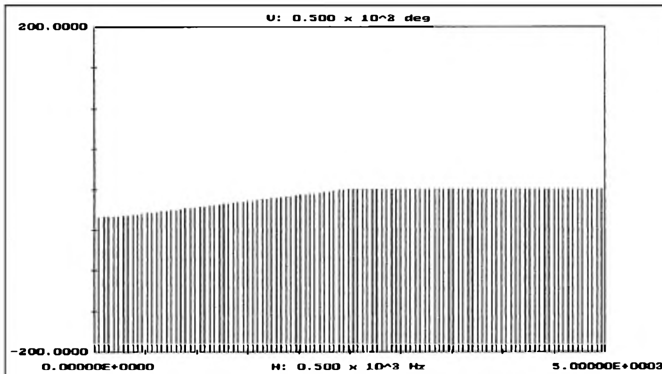



FIGURE 25: Phase correction applied to the ESL phase response of *Fig. 24* to linearize the phase in the crossover region.

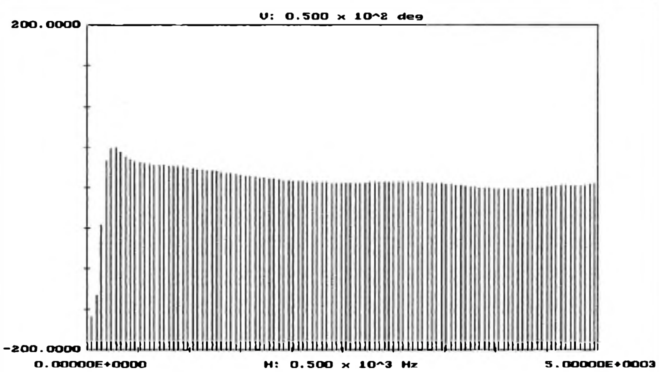



FIGURE 24: ESL phase response, also derived from the pulse response of *Fig. 22.*

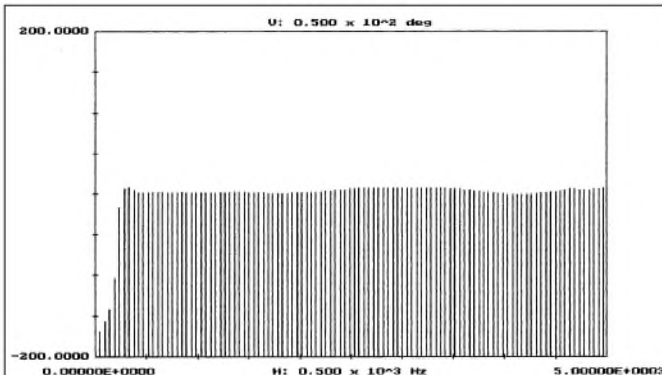



FIGURE 26: Measurement of ESL phase response with microphone 6′ away from panel after application of phase correction from *Fig. 25.*

In this circuit, the inductors and capacitors determine not only the resonant frequencies, but also the width of the traps; making L larger and C smaller (while maintaining a constant product) makes the notch filter narrower. At this impedance level, the required inductance value of 4H is quite large. An advantage of the Virtual Crossover is that it is not necessary to wind such a huge coil; all you need do is type the inductance value into the simulation.

## IDEAL OP AMPS

I should emphasize that the op amps in *Fig. 20* are not real ones; rather, they represent ideal op amps used in the impulse-response calculation. These ideal amplifiers have infinite input impedance and a selectable gain that is constant with frequency. I chose for them a gain of $10^4$.

*Figure 21* shows the response when the woofer trap stage of *Fig. 20* is included. There is still a substantial dip in the response around 1200Hz, but since the third-order low-pass filter is down by 23dB at this frequency, I decided not to try to compensate for it; I was satisfied just to tame the peaks in the response.

The next step is to measure the ESL panel. On the low end, a phenomenon that Roger Sanders calls "phase cancellation" dominates the ESL frequency response. In a dipole speaker at low frequencies, the back wave diffracts around the speaker and blends with the forward wave, and since the two are out of phase, a reduction in sound output occurs. This effect shows up most prominently in the far field, so it is important to place the microphone far enough away from the ESL; I took the measurements with the microphone about 6′ away from the ESL panel for this design.

*Figure 22* shows the ESL pulse response I measured using an input filtered pulse designed for measurement out to 5kHz; I truncated the response to the right of the vertical line before calculating the FFT. I will not present the truncated time waveforms with the remainder of the measurement results in this article, but I wanted to show one example in order to clarify the method. This technique is a very powerful way to approximate measurements in an anechoic chamber, but it has its limitations. In particular, it is not always clear where the primary speaker response ends and room effects begin. In *Fig. 22* it is obvious that the echo that occurs around 12ms is due to a reflection from the room, but it is not as clear whether the signals immediately following the truncation point are due to room effects or perhaps interactions with the speaker cabinet.

*Figure 23* shows the amplitude response of the ESL panel out to 5kHz based on the truncated waveform of *Fig. 22*. You can see that the ESL response is flat down to around 1500Hz, but below this frequency it drops off due to the phase cancellation effect. It is important to take this effect into account in the crossover design; otherwise, the resulting system will sound thin and unnatural. I chose to introduce a frequency compensation stage for the ESL, providing 14dB boost with corner frequencies of 1250Hz and 250Hz. The first stage in the lower half of *Fig. 20* shows the circuit I used to equalize the low-frequency ESL response.

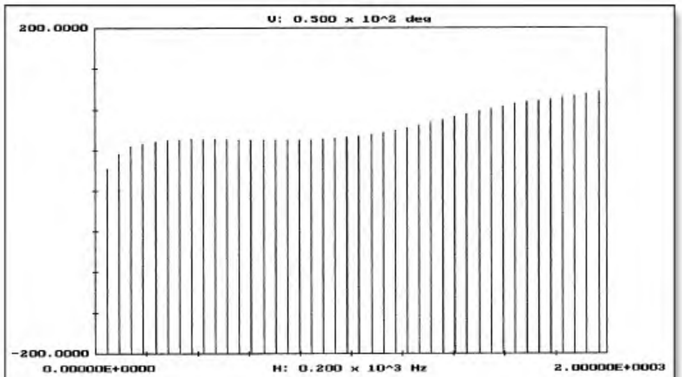



**FIGURE 27: TL phase response before compensation measured 6′ from the speaker.**

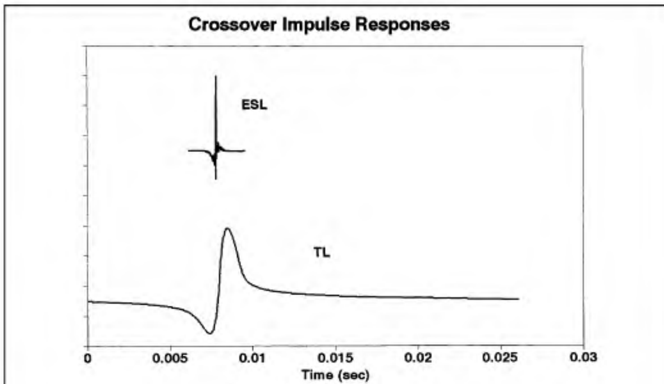



**FIGURE 29: Impulse responses used to characterize the ESL and TL filters in the Virtual Crossover simulation.**

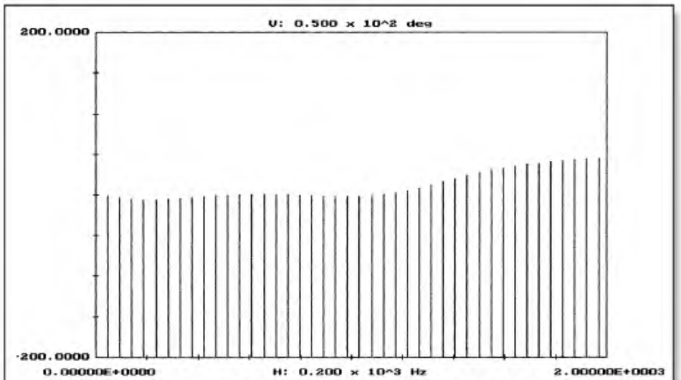



**FIGURE 28: Measured TL phase response after application of constant –60° phase compensation.**

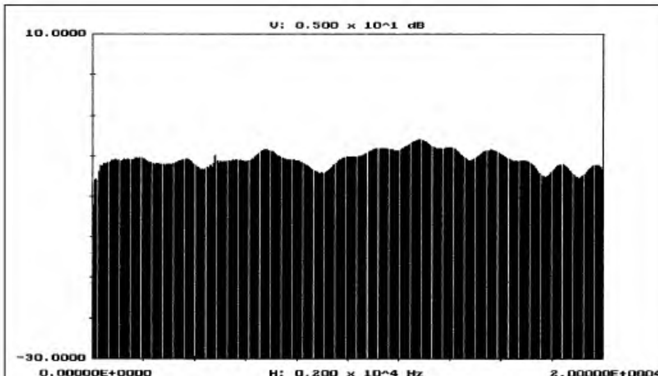



**FIGURE 30: Measurement of combined ESL/TL system out to 20kHz, with microphone about 6′ away from the speakers.**

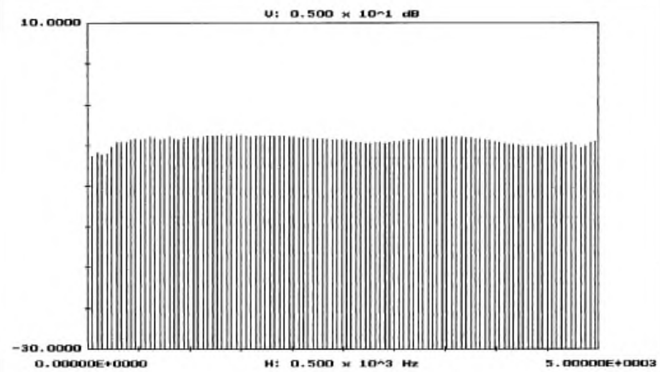



**FIGURE 31: Measurement of combined ESL/TL system out to 5kHz, with microphone about 6′ from speakers.**

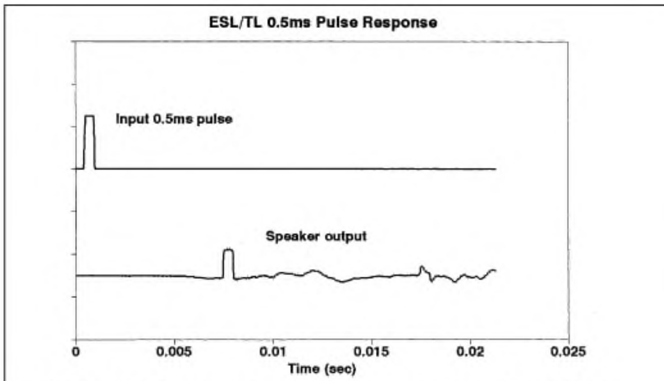



**FIGURE 33: 0.5ms square pulse response of combined ESL/TL system, microphone 6′ from speaker.**

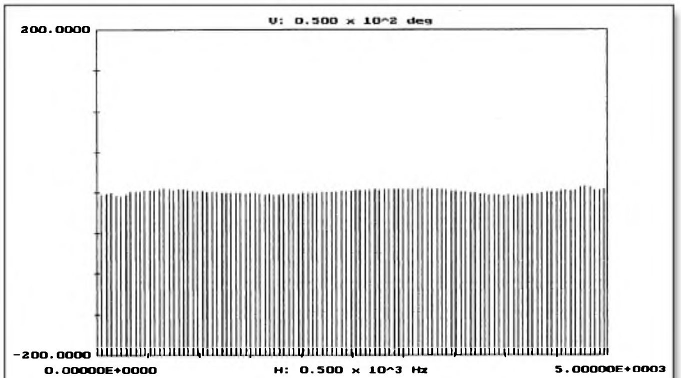



**FIGURE 32: Phase response of combined ESL/TL system out to 5kHz.**

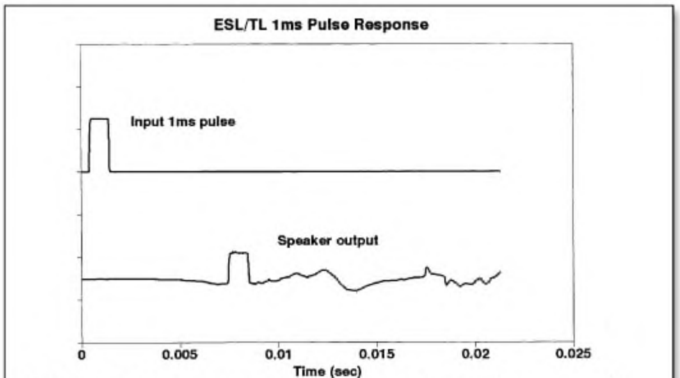



**FIGURE 34: 1ms square-pulse response of ESL/TL system, with microphone 6′ from speaker.**

In addition to the amplitude response of the drivers, the phase response is also very important. Ideally, you would like the drivers to have linear phase response, with the values and slopes of the phase responses matched at the crossover points. Using digital signal processing techniques it is possible to equalize the phase responses of the drivers separately from the amplitude responses.

*Figure 24* shows the ESL phase response derived from the same pulse information as *Fig. 23*. Phase response curves depend upon the choice of zero time reference—changing the time origin adds a linear phase component to the curve. In *Fig. 24*, I chose the zero time reference to flatten the phase response above about 2500Hz.

In this range the phase response is very flat and near 0°, which is the desired situation; however, the phase response below 2500Hz increases as frequency decreases. To compensate, I introduced ESL phase compensation, which is easily accomplished in the digital domain. Transform the pulse response to the frequency domain by taking the DFT (Discrete Fourier Transform), adjust the phase in the frequency domain as desired, and then transform the new frequency information back to the time domain using the inverse DFT transform. I have written programs, which I will post on my website, called DFTRI and PHCOMP, that can be used to apply phase compensation.

*Figure 25* shows the phase equalization applied to the ESL phase response of *Fig. 24* in order to extend the region of linear phase to below the crossover frequency. I used a linear phase correction term, starting with 0° at 2500Hz and reaching a maximum negative correction of –33.75° at 250Hz, one octave below the crossover frequency. The implementation of this phase response correction term is indicated by the box labeled "phase compensation" in *Fig. 20*. *Figure 26* shows a re-measurement of the ESL phase response after application of the phase compensation in *Fig. 25*.

*Figure 27* shows the measured phase response of the transmission line, also with the microphone about 6′ away from the speaker. This response is not as easily measured as the phase response for the ESL—the woofer radiates sound more omnidirectionally than the ESL, so it is more difficult to truncate the pulse response in order to remove room interactions. I derived *Figure 27* from a truncated time response believed to be a good indication of the phase of the early arrival sound from the woofer.

In *Fig. 27* you can see that the phase in the crossover region is fairly flat; however, it is not flat near 0°—rather it assumes a value around 60°. To correct for this phase offset, I decided to subtract a constant 60° phase from the entire TL phase response. *Figure 28* shows a measurement of the TL phase response after this phase compensation was introduced, illustrating that the phase compensation does indeed bring the phase curve down to 0° as intended.

There is, however, a disadvantage to this choice of phase compensation—it spreads the TL filter impulse response out much more in time. This is evident in the plots of the final impulse re-

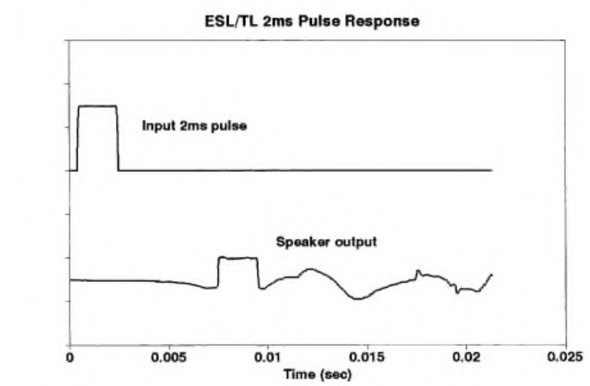



**FIGURE 35: 2ms square pulse response of ESL/TL system, with microphone 6′ from speaker.**

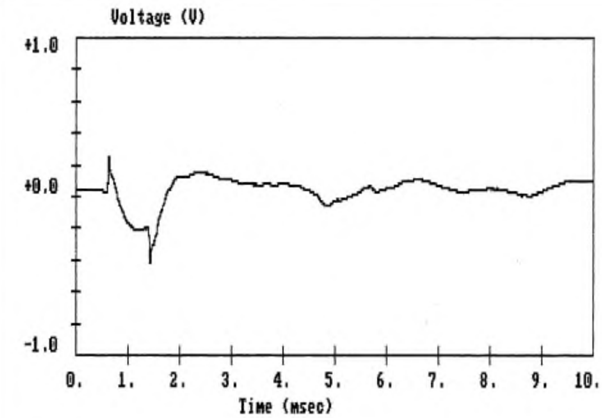



**FIGURE 36: 1ms square pulse response of combined ESL/TL system previously obtained, using real op amp circuits.**

sponses presented later in the article. I do not yet fully understand how to select the optimum phase compensation profile, and it is quite possible that subtracting a constant 60° from the TL phase is not the best choice.

I should mention at this point that you must be careful about applying modifications to the frequency-response characteristics of the crossover filters, both in amplitude and phase. After making the modifications in the frequency domain, you must transform the results back to a time-domain impulse response that is required to be real, i.e., it must not contain an imaginary part.

You can accomplish this if you adhere to two simple rules. First, any phase changes must be odd functions about the zero frequency (DC) point, and second, any amplitude changes must be even functions about this point; at DC, the phase correction must be zero. If you are familiar with Discrete Fourier Transform theory, you know that frequencies greater than the Nyquist folding frequency (roughly half the maximum frequency) may be equivalently considered as negative frequency values that produce a frequency range symmetric about zero frequency.

## COMBINING THE DRIVERS

I used third-order low-pass and high-pass filters for this system, as recommended in Roger Sanders' article. One approach to implementing these filters would be to start with op-amp implementations of them, and then to remove the net phase shift from the combined output of the filters. However, I chose a more direct approach, one which also illustrates a unique feature of the Virtual Crossover. Instead of starting with real circuits, you can simply specify the ideal third-order frequency responses that you wish, and then transform them directly into impulse responses.

For the low-pass filter in the TL crossover circuit, I chose the following response:

$$A_{lp}(f) = \frac{1}{1+\left(\frac{f}{f_0}\right)^3}, \qquad (10)$$

where $A_{lp}$ is the desired amplitude response of the low-pass filter as a function of frequency f. In equation 10, $f_0$ is the crossover frequency, which for this case is 500Hz. The corresponding high-pass response for the ESL filter is given by:

$$A_{hp}(f) = \frac{\left(\frac{f}{f_0}\right)^3}{1+\left(\frac{f}{f_0}\right)^3}. \quad (11)$$

Note that there is no imaginary part in equations 10 or 11, so these filters have no phase shift. Note also that at the crossover frequency each response is down by exactly half, so that the responses add up exactly to one, which in fact also occurs at any other frequency.

I wrote programs called THDLP and THDHP to calculate the impulse responses associated with the third-order filters specified by equations 10 and 11. For the high-pass filter of equation 11, it is also necessary to band-limit the response, since this function as presented is not band-limited; I did this by convolving the high-pass impulse response with the band-limited impulse that I presented in Part 1 on the theory of the Virtual Crossover. I accomplished this convolution with a program I wrote called SIMULATE.

## IMPULSE RESPONSES

I used these filter impulse responses as inputs to the equalization circuits of *Fig. 20* to obtain the overall impulse responses of the crossover filters. I then adjusted the phase responses as described previously to obtain the final impulse responses. *Figure 29* shows the resulting impulse responses used to characterize the ESL (top trace) and TL (bottom trace) filters in the Virtual Crossover. These waveforms have been truncated in time on both sides of the peak value, in order to reduce the calculations that must be performed in each 48kHz interval by as much as possible.

The time durations of the impulse responses are quite different; the TL filter response is quite long, due in large part to the 60° applied phase compensation. Fortunately, since the TL impulse is band-limited (due to the 500Hz low-pass filter), a trick can reduce the number of calculations required for this impulse response.

I designed a filter using the ERLPF program to band-limit the input signal to 4kHz. This band-limiting filter contains 109 points, and I perform a convolution of this filter with the input signal to produce an input that is band-limited at 4kHz. You may then calculate the convolution integral of this band-limited input with the TL impulse response using only every sixth point, i.e., at a sampling rate reduced by a factor of 6 from the 48kHz sampling rate. I should also mention that the waveforms in *Fig. 29* are not shown to scale—since the ESL circuit is high-pass, the peak of its impulse response is about 30 times the peak of the TL impulse response.

*Figure 30* shows a measurement I made of the combined ESL/TL speaker system out to 20kHz with the microphone placed about 6′ away from the speakers. I implemented the entire crossover circuit using the impulse responses in *Fig. 29*. I truncated the measurement after a few milliseconds, so you should regard it as depicting the early arrival sound from the speakers. *Figure 31* shows the overall response out to 5kHz (using a wider input pulse for the measurement), which was also truncated in time to avoid the effects of room interactions. At least for the early arrival sound, the response is quite flat throughout the crossover region.

*Figure 32* shows the phase response of the combined ESL/TL speaker system, from the same measurement used for the magnitude response in *Fig. 31*. You can see that the phase compensation introduced in both the ESL and TL crossover filters achieved flat phase response over a wide frequency band.

## BAND-LIMITED PULSES

Finally, I would like to show square pulse responses for the hybrid ESL/TL speaker system. Actually, I could not use square pulses, because a perfectly square pulse is not band-limited. Instead, I constructed pulses with slightly rounded edges that were band-limited using the ERLPF program that I discussed in Part 1. This waveform retains the basic shape of a square pulse, but has negligible frequency content at and above 20kHz. For these measurements, I located the microphone about 6′ from the speakers.

The top trace of *Fig. 33* shows the band-limited 0.5ms input pulse I used to test the speakers, and the lower trace shows the response of the ESL/TL system to this pulse. Following the primary speaker response you can see the effects of room interactions. I can't say exactly where the speaker response ends and the room effects begin, since I was not able to carry out the measurement in an anechoic chamber. *Figures 34* and *35* present the input pulses and pulse response waveforms for wider pulses of 1ms and 2ms duration. For pulse widths greater than 2ms there is no clear demarcation between speaker and room response at a measuring distance of 6′ from the speaker.

There is an effect that becomes more noticeable in *Figs. 33–35* as pulse width increases—you can see a "precursor" before the primary speaker response, where the response drops to a lower value. I do not entirely understand the reason for this effect or how to eliminate it. At this point I will just remark that since the filter impulse responses of *Fig. 29* have a long tail preceding the response peak, particularly for the TL filter, this type of behavior is possible.

For comparison of these waveforms with what I was able to obtain previously, *Fig. 36* shows the 1ms pulse response I obtained before, using an active crossover with real op-amp circuits. The phase shift introduced by the third-order low-pass and high-pass filters results in the low-frequency and high-frequency portions of the output being essentially out of phase. (Actually, the polarities of signals to the ESL and TL are reversed in this case; if you don't reverse the polarities, the square-pulse response is even worse, because there is more overall phase shift throughout the crossover region). The amplitude response for such a system can be quite flat; however, the phase response is very nonlinear.

## THE SOUND

It was not my intention to discuss at length the merits of the hybrid ESL/TL speaker in this article, but I will make a few comments. These speakers have great potential, but they are also very demanding and require careful implementation to realize their potential. Crossover design is critical, and the ESLs are highly directional, so careful placement is important.

Setup and measurement of this system is difficult due to the fact that the woofers tend to radiate sound omnidirectionally, while the ESL panels focus the sound in the forward direction. Despite these shortcomings, if everything is done well these speakers can sound absolutely stunning.

I have worked on several versions of crossovers for the ESL/TL system, using both first- and third-order crossover filters. First-order filters are not practical, because the drivers do not cut off fast enough, and the ESL panels cannot tolerate low-frequency signals. Since standard third-order filters introduce significant phase shifts, it has so far not been possible for me to obtain good transient response from this system.

Using the crossover presented in this article, these speakers sound more natural to me than they have ever sounded before. I have the impression that I am hearing exactly what is being played through them, with very little coloration.

## LIMITATIONS OF THE VIRTUAL CROSSOVER

Although the Virtual Crossover is a very useful device that allows you to do things that are not possible with real crossovers, it has some significant limitations. Probably the biggest limitation for this device is the amount of computation that it can carry out. It must perform a complete calculation of the convolution integral (in discrete form) for each channel and crossover filter during every 48kHz sampling interval.

I think a safe upper limit for the total number of points you can calculate with this device is about 960 points, or 480 points per channel. It is possible to calculate fairly long filter impulse responses if they are band-limited, but there is still an upper limit that can easily be reached, particularly if you are using very low crossover frequencies, such as might be used in a crossover with a subwoofer. If you are thinking of using this device, I would first take a look at the typical impulse response waveforms that you would need to use for your system, to see if the DSPs can handle them. To do this, you are welcome to use the software that I provide on my website.

Another limitation arises from the rated output of the AD1819A Codecs, which are capable of producing about 1.41V peak analog output before clipping. My power amplifiers have a gain of about 20, which means that I would be able to deliver about 50W output to an 8Ω load unless further amplification were introduced between the Codec output and power amplifier. This output level may not be sufficient for your application.

Finally, the distortion level rating for the AD1819A Codecs may not be sufficiently low for some audiophiles. The total harmonic distortion for both the A/D and D/A sections of this device is rated at 0.02%.

## CONCLUSION

I hope I have demonstrated that working in the digital domain opens up a variety of new possibilities for designing speaker crossovers. I have presented one application in the design of an active crossover, which eliminates the phase shifts characteristic of third-order analog filters. While my intention in this application was to replace the real crossover with a virtual one, this device can also be used to optimize the real crossovers that will eventually replace it.

As DSP techniques become more powerful and affordable, I believe that these approaches to speaker design will become more widespread. ❖

### REFERENCE


1. Sanders, Roger R., "An Electrostatic Speaker System—Parts I, II, and III" *SB* 2/80, 3/80, and 4/80. See also "A Compact Integrated Electrostatic/TL" *SB* 1/90, 2/90, and 3/90.

2. Mensicus Audio Group, Inc, 4669 S. Division, Wyoming, MI 49548 www.MENSICUSAUDIO.com.


### RESOURCES

I have posted the software that was used in the development of the Virtual Crossover on my website, www.usol.com/~rkm/audio. Questions regarding the software or these articles can be directed to my e-mail address, rkm@usol.com.

# Glass Shard

## Lafayette Preamp

For the last five years, I've been trying out commercial phono stages, hoping to find one that was easy to build and, of course, the best sounding. In many of the integrated amps that had built-in preamps, I needed to study the switching circuit to determine the proper RIAA compensation circuit, which was different in each position of the selector switch. In many amps this proved to be just too difficult.

I tried many of the simpler designs, which, for the most part, didn't have the quality I was looking for. When I tried the preamp used in the Lafayette KT-250/LA-250 amp, I was amazed at what I discovered! This preamp sounded fantastic and was easy to build.

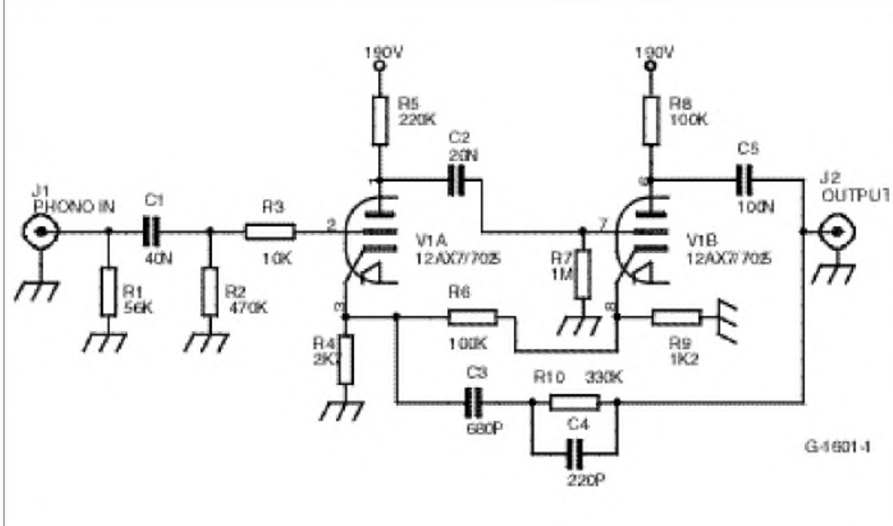



**FIGURE 1: Phono stage from Lafayette KT-250/LA-250.**

### EXCELLENT CHOICE

As with most stages, this circuit uses one 12AX7 double triode per channel and features positive feedback from the cathode of the second triode to the cathode of the first triode. This method is also used in the Scott 299's phono stage. According to Charles Kittleson in his *Vacuum Tube Valley* article on the Scott 299, this design "...increases overall loop gain and produces a rich musical midrange with good response on the frequency extremes." By listening to this preamp, I say he's right on the mark.

This preamp is an excellent choice for those who don't have hundreds, or thousands to spend on today's preamps. Further, you don't need to use high-priced parts. I used salvaged parts, including tubes, and I'm very satisfied with the sound. *Figure 1* is the circuit. Any power source supplying 190V DC and a DC filament source should be adequate. ❖

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# Kora Explorer 90SI

***Reviewed by Charles Hansen***

**PHOTO 1: Explorer front view.**


*Kora Electronic Concept, 81 Chemin de Fenouillet, 31200 Toulouse, France, +33 (0) 5-62-72-43-13, FAX +33 (0) 5-62-72-43-14, kora@wanadoo.fr, www.kora.net. Dealer list is at www.kora.net/distribu.html. Price: $750 US. Dimensions: 420mm W × 260mm D × 90mm H; net weight: 7kg, 15.4 lbs. Warranty two years, tubes 90 days.*

The Kora Explorer is the first tube/MOSFET hybrid in the Kora product line, and is rated for 60W RMS into 8Ω and 90W into 4Ω. *Photo 1* shows the front panel, which has an operate switch, two LEDs, an Alps volume control, pushbutton switches for the tape loop and external A/V processor, and a five-position Alps rotary switch. The latter selects any of five line-level inputs (CD, tuner, and three other line inputs). The plastic logo plate above the LEDs has a clear section that allows one of the ECC81 tubes to show through.

The amplifier is nicely constructed of heavy gauge steel, with a 3mm-thick black anodized aluminum front plate. The top and bottom covers have cooling slots for the MOSFET heatsinks. The top is held in place with eight flat-head cap screws and is rattle-free.

The rear panel (*Photo 2*) has the IEC power receptacle with integral fuse and power switch, 16 high-quality gold-plated Teflon™-insulated RCA jacks, and two pairs of high-quality gold-plated speaker binding posts, which are located on US 0.75″ spacings, so you can use cables with dual banana plugs.

*Photo 3* shows the amplifier with the cover removed. The left side is dominated by a hefty custom toroidal power transformer with a 230V AC, a 6V AC, and two 26V AC secondary windings. There is very little hand wiring in this unit, with all the electronic parts on a single large PC board.

High-quality components are used throughout the amplifier. The elegant epoxy PC board uses 1% metal-film resistors, polypropylene caps, ceramic tube sockets, and Alps pots and switches. Most chassis-mounted components (input jacks, volume control, LEDs, and switches) are soldered directly to the PC board.

Each of the output MOSFETs is mounted on its own heatsink, with a driver FET in close proximity. A trimpot located near each heatsink provides adjustments for output device bias and DC offset. Speaker protection fuses are mounted between the heatsinks in green plastic holders.

The unit is furnished with a heavy power cord. The power transformer primary is factory-connected for either 115V or 230V mains. The third pin of the IEC power receptacle is connected to the chassis. The Explorer sits on four small elastomer dots, which makes finger space under the unit a bit tight. However, the amplifier is not all that heavy and is easy to lift.

## TOPOLOGY

A schematic was not furnished with the unit, but the ten-page user's manual has an explanation of the circuitry in the Design Philosophy section. During the conception period the design philosophy called for each component to be given the role that suits it best—voltage amplification for the tubes and current amplification for the MOSFET transistors.

The unit was furnished with two matched Philips JAN ECC81/12AT7 tubes. A "tweak" section in the manual describes the difference in sound if you substitute 12AX7 or 12AV7 tubes.

When you operate the AC power switch on the rear of the unit, the red LED immediately comes on. The operate switch on the left front panel controls only the MOSFET power-supply relay, whose driver transistor also illuminates the orange LED. When you release the

**PHOTO 2: Explorer rear view.**

operate switch, the power amplifier turns off (standby), while the tube sections of the amplifier remain energized and ready for action.

The selected stereo input is connected to the grids of the first 12AT7 through the tape monitor switch. The plates are then capacitively coupled to the volume control. The tape out jacks receive the selected input signal directly from the selector switch, with no active buffer or intervening resistors.

The volume-control wipers are connected to the grids of the second 12AT7 through the processor switch. When you press the processor switch, the A/V stereo input is connected to the grids of the second tube, bypassing the input selector, tape loop, and volume control.

The second 12AT7 serves as the voltage amplifier for the MOSFET output stages. Each plate is R-C coupled to the gates of complementary-pair driver MOSFETs. Trimpots in their gate circuits set

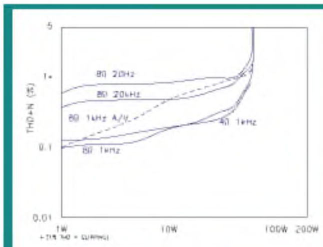



FIGURE 1: THD+N versus output.

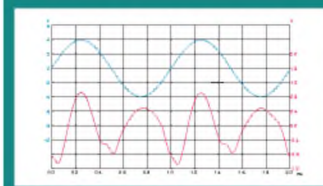


FIGURE 2: Residual distortion, line input.

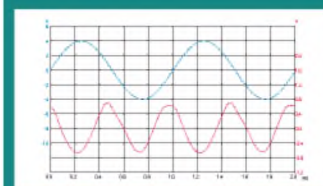


FIGURE 3: Residual distortion, A/V input.

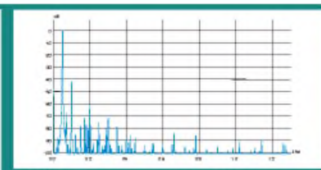


FIGURE 4: Spectrum of 50Hz sine wave, line input.

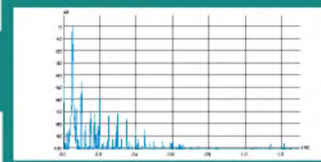


FIGURE 5: Spectrum of 50Hz sine wave, A/V input.

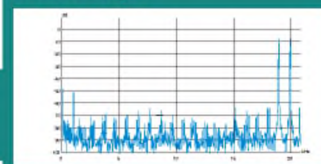


FIGURE 6: Spectrum of intermodulation distortion.

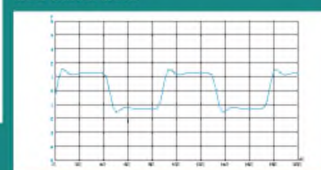


FIGURE 7: Square-wave response, 10kHz 2.5V p-p 8Ω in parallel with 2μF.

## TABLE 1
### MEASURED PERFORMANCE

| PARAMETER | MANUFACTURER'S RATING | MEASURED RESULTS |
|---|---|---|
| Power output (RMS) | 2 × 60W, 8Ω | 61W 8Ω, 1% THD |
| | 2 × 90W, 4Ω | 40W 4Ω, fuse limited |
| Slew rate | 1μs | |
| Signal to noise ratio | −100dB | |
| Frequency response | 7Hz–40kHz, −3dB | 3Hz–47.5kHz, −3dB |
| Total harmonic distortion | N/S | 0.95%, 60W, 8Ω |
| IMD—CCIF (19 + 20kHz) | N/S | 0.28% CCIF, 8Ω |
| MIM (9 + 10.05 + 20kHz) | 0.075% MIM, 8Ω | |
| Line input impedance | 100k, 330mV | 98k2 |
| A/V input impedance | 47k, 1.4V | 47k6 |
| Noise | 0.8mV | 1.2mV |
| Gain | N/S | 38dB line input, 8Ω |
| | | 19dB A/V input, 8Ω |
| Power requirements | 200W maximum | |
| Output impedance | N/S | 0.69Ω 1kHz |
| | | 0.71Ω 20kHz |

**PHOTO 3: Explorer interior view.**

the bias and DC offset in the output MOSFETs.

Since MOSFETs require about 6V gate-source voltage, the output devices have diode-capacitor peaking networks in their gate drive circuits that hold the gate voltage at the peaks of the power supply to increase gate drive. The output MOSFETs in each channel are also a complementary pair—IRFP150 and IRFP9140—which are connected to the speaker binding posts through source resistors and a speaker protection fuse. Overall feedback is a low 6dB.

The power supply reflects the complexity of a hybrid design. The MOSFETs receive power from the 26-0-26V AC windings, full-wave rectified to about ±36V DC. The 10,000µF caps provide power-supply filtering for each rail. The operate switch time-delay relay connects the AC windings to the rectifier.

The 230V AC winding is full-wave rectified and filtered to about +340V DC. This high voltage goes to a source-follower MOSFET B+ regulator, whose output voltage is determined by a zener reference string. The gate circuit has an R-C startup time delay that allows the tubes two minutes to heat up before B+ is applied.

The 6V AC winding powers the 12AT7 heaters. This winding is also rectified to power the LEDs and the MOSFET power-supply time-delay relay.

## MEASUREMENTS

I operated the Explorer at 10W per channel into 8Ω for one hour. There was no speaker noise during power-up or shutdown. At full volume, with my ear against the speaker, there was a fair amount of 120Hz buzz that varied with the volume-control setting.

I set the volume level so the line-input gain equaled that of the A/V input. The line-input noise was quite a bit higher than that of the A/V input when I toggled between them with the processor switch. The scope waveform showed high levels of diode commutation spikes on the 120Hz ripple. Despite all this, the amplifier was quiet from my listening position.

The Explorer uses two cascade tube stages in its design. The five line inputs and the tape inputs are amplified by both tubes and have normal polarity. However, the A/V processor input is connected to the grid of the second tube, and this inverts the input polarity as seen at the speakers.

Overall line-input gain for 2.83V RMS output with the volume control at maximum is 37dB into 4Ω, and 38dB into 8Ω. The A/V input had a fixed gain of 18.3dB into 4Ω, and 19dB into 8Ω. The line-input gain equaled the A/V input gain when the volume control was one dot below 12 o'clock.

Line-input impedance for both channels measured 98k2 at 1kHz, independent of the volume-control setting, while the DC resistance at the selected input jack was over 1M. This suggests there is negative feedback around the first preamplifier tube, since there are no input coupling capacitors. Similarly, the A/V input impedance at 1kHz, with the processor switch on, measured 47k6, with a DC resistance of 1M.

The amplifier output impedance was 0.69Ω at 1kHz, and 0.71Ω at 20kHz, which is quite flat for an amplifier with only 6dB of feedback. The amplifier frequency response is not likely to be affected by speaker impedance variations.

The frequency response for the Explorer was within +0, −3dB from 3Hz to 47.5kHz, with 0dBr defined as 2.83V RMS at 1kHz into 8Ω. It was −1dB at 4Hz and 18.3kHz. With a 4Ω load, it was a bit flatter at the top end, being −1dB down at 20kHz, with the −3dB point essentially the same. The HF gain rolled off smoothly to −8dB at 100kHz.

When I connected a load of 8Ω paralleled with a 2µF cap, the HF −1dB point moved out to 36.5kHz, with −3dB at 65kHz. There was no peaking at higher frequencies with the capacitive load, and no evidence of ringing or instability. The amplifier handled the IHF simulated speaker load, which has an impedance peak at 50Hz, very well. The response was only 0.4dB higher at this frequency.

Channel separation measured −73dB at 100Hz, −53dB at 1kHz, and −33dB at 10kHz L-R. The input signal traces run right next to each other for the full depth of the PC board, and the selector switch is fairly small, so there is ample opportunity for capacitive coupling between channels. Volume-control tracking across the entire rotational range was excellent, as you can usually expect from Alps controls.

The distortion was a bit higher for the left channel, so I present its test data here and summarize it in *Table 1*. THD+N versus frequency (line input) showed a fairly high level at 20Hz, just under 1% up to 60W, and pretty much independent of power level or load. THD dropped with frequency up to about 1kHz, then increased again gradually up to 20kHz. With the A/V input, the THD+N at 10W into 8Ω was essentially constant at about 0.4% from 20Hz to 20kHz.

*Figure 1* shows THD+N versus output power using the line inputs: into 8Ω at 20Hz, 8Ω at 20kHz, 4Ω at 1kHz, and 8Ω at 1kHz (solid lines top to bottom at 5W). The dashed line is the A/V input at 1kHz and 8Ω. The 4Ω and 8Ω 1kHz lines cross at about 13W.

The clipping level at the line input where THD+N reached 1% occurred at 61W into 8Ω. At 65W into 4Ω, the left-channel-speaker protection fuse blew. I opened the cover and removed a T2A fuse (two amps time delay), which would certainly not allow the full-rated 90W at 4Ω. I was surprised it held at 60W, which is almost 3.9A RMS into 4Ω, but the Littelfuse type 219 time-current curve (www.littelfuse.com) shows a maximum of two minutes is available at 210% of rating. The replacement fuse let go at 45W (3.35A), so I have no THD data above 60W—highly variable, these fuses!

There is no mention of the fuses or their ratings in the manual. I reported the fuse openings to Kora, thinking perhaps that they used smaller fuses for the trade show where the amplifier was shipped to me. However, T2A is the proper rating, since Kora wishes to keep a dynamic reserve of 6dB at 90W (around 23W average). My tests show the output stage is capable of much more power. If you have problems because your speaker impedance is too low or you listen to high volume "compressed" music, you can increase the fuse to a maximum of 4A. I decided to finish my tests with the 2A fuses, since that is the value the factory ships with the Explorer.

With my 8Ω test loads, I could get the full-rated 60W at 1kHz,

**Reviewed by Duncan and Nancy MacArthur**

The Explorer is a relatively uncomplicated integrated amplifier capable of producing a musical sound with most recordings. Few amplifiers we've reviewed have inspired us to listen to CD after CD just to hear the music. To our surprise (given its modest price tag), the Explorer belongs to this small group. This amplifier produces strong, well-defined bass and clean, extended highs.

The biggest problem that we had with the Explorer was inconsistency. Don't misunderstand—this amplifier always sounded good. But occasionally it sounded very good indeed. We couldn't correlate this change with any external effect: power fluctuations remain a possibility.

On the downside, the Explorer sometimes sounded slightly hard and "glassy" at high volumes. Most other defects were subtractive rather than additive in nature. The imaging was good but not outstanding, and the Explorer lacked that last sparkle of "aliveness" that characterizes the best amplifiers (albeit at a much higher price).

### AESTHETICS AND OPERATION

Like most components nowadays, the Explorer is a black box. While the Explorer is finished well enough that it won't detract from the rest of your equipment, it isn't a component that demands to be shown off.

One of the preamplifier tubes is mounted behind a small clear window in the front panel. Duncan sees this design as a marketing ploy meant to "prove" that there are tubes in the amplifier. He can't imagine that any of the signal paths have been improved by placing this tube at the front of the chassis. However, he appreciates seeing that the filament is operating. Nancy, on the other hand, likes the tube window. She thinks it looks like the window of a glowing furnace room in a doll's house.

In the process of changing the output fuses, we had an opportunity to examine the construction quality of the Explorer. The top cover is thick, well painted, and attached with machine screws rather than the more common (and cheaper) sheet-metal screws. All the components, except the power transformer and output jacks, are mounted on a single large printed-circuit board.

The input connectors are mounted directly to this PCB, but, because the source selector switch is mounted directly behind the front panel with no shaft extension, this amplifier has multiple long PCB traces at the amplifier input. All the components are securely mounted, with the exception of the four heatsinks, which are supported only by the leads of the output MOSFETs. Supporting the heatsinks in this fashion cannot be good for the long-term reliability of the amplifier.

We experienced some quirks in interfacing the Explorer into our system. The speaker connectors are large "thumbscrew type" binding posts that ensure good connections but cause a couple of minor problems. The left-channel output connectors are located so that the speaker wire must pass directly over the input jacks. If your speaker cables are relatively stiff or you use large connectors on the end of your speaker cable, then these connectors and cable will block access to the A/V inputs and perhaps the tape inputs and outputs. Large connectors must be insulated to avoid short circuits to the shield connection on the A/V jacks.

Also, although we use oversize "spade" connectors for speaker attachment, and these spades fit around every speaker output post we've encountered, the posts on the Kora are marginally too large for our connectors. To achieve proper connection we needed to use some force.

Finally, as mentioned in the instruction manual, the front panel "operate" switch does not completely remove power from the Kora. Reaching behind the amplifier is required in order to turn it off fully. Although you may be tempted to use only the front panel switch and leave the Explorer in continuous standby mode, we do not recommend this type of operation. We noticed a slight negative correlation between the length of time the filament had been powered and the overall quality of sound. This effect only occurred after several weeks of continuous filament power, so don't worry about keeping it turned on all day.

Our sample of the Kora Explorer 90SI had been shipped several times in the same box prior to its arrival here. The packaging, which consisted of a single-layer cardboard box with several internal cardboard spacers, was insufficient to protect the amplifier throughout this process. In particular, one corner of the front panel was slightly damaged.

### A TALE OF TWO FUSES

During our initial auditions, the Explorer sounded quite musical, but the imaging bordered on unacceptable on many recordings. After an e-mail exchange with Jeff Starrs at Kora, we opened the Explorer and examined the output fuses. One channel used a 2A fast-acting fuse and the other a 2A slow-blow version.

Further discussion with Jeff established that slow-blow fuses should have been used in both positions. Using a pair of 2A slow-blow fuses greatly improved the imaging. Unfortunately, some of the musical magic that had been present earlier was lost.

With slow-blow output fuses, the Explorer sounded slightly "rounded off" in all respects. The transient response was good but not great. Ditto the dynamics. The highs seemed slightly rolled-off, which was a boon with poorly recorded CDs but did nothing for the sound of well-recorded ones.

We made another trip to Radio Shack to purchase fuses. (As you might expect, the fuses in the Explorer are 5mm by 20mm metric packages.) Using two 2A fast-acting fuses in the Explorer's outputs produced some sonic improvement. The sound was more musical and the imaging at least as good as that achieved by using a pair of slow-blow fuses.

If you own the Kora Explorer or contemplate buying one, consider purchasing a pair of fast-acting fuses. The difference is significant. Replacement fuses are clearly the best $1.99 (for four fuses) investment you could make.

An exception to this advice might occur if you regularly play the Explorer at high volume with inefficient speakers. The slow-blow fuses will allow a higher peak output. We never blew the fast-acting fuses, even when auditioning at higher than normal levels. In addition, you could use the slow-blow fuses as a subtle "tone control" to tame the obnoxious highs on some CDs. The remainder of this review (unless explicitly stated otherwise) refers to the Explorer with fast-acting fuses installed.

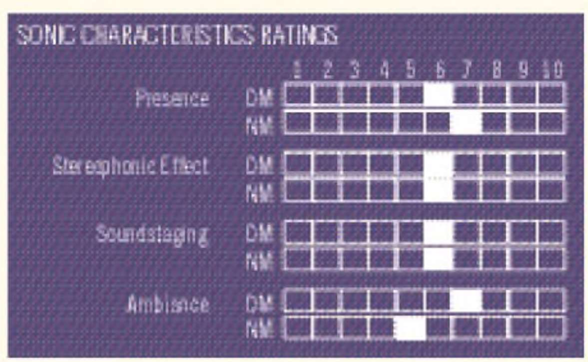
### DETAILED LISTENING RESULTS

All our auditioning was done using Genesis 400 speakers. These three-way floor-standing systems (not bi-amplified) are rated at 89dB/W efficiency but are 4W designs with a fairly difficult impedance curve. We compared the Kora directly with an all-tube integrated amplifier, the Manley Stingray. This comparison is patently unfair because the Manley costs three times as much as the Kora. And no, the Kora doesn't surpass the Manley, but it performs remarkably well for an amplifier in its price range.

Although the sound of the Explorer did not improve noticeably during our month-long listening period, we wouldn't draw any conclusions about required break-in. Apparently our sample had been to CES prior to coming to us and would have had ample opportunity to break in there. As mentioned above, we do not recommend leaving the tube filaments constantly powered; we routinely played the amplifier for at least one hour before doing any serious listening.

We listened carefully to several tracks on the *Hi-Fi News and Record Review* disk III (track 2: Parry's "Jerusalem," track 4: Vivaldi's trumpet concerto, tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: Rio Napo RSS demo). We also listened to a wide variety of other CDs.

The Explorer 90 sounded good with music ranging from The Modern Jazz Quartet (*Blues on Bach*, Atlantic compact disk, 1652-2) through Jethro Tull (*Aqualung*, DCC, GZS-1105) to Beethoven (*The Complete Piano Sonatas*, Robert Silverman, Orpheum Masters KSP 830). Although the Explorer did sound natural as evidenced by the clear reproduction of both the voices and harpsichord on "Welcome, Welcome Glorious Morn," the good sound of the Explorer wasn't completely attributable to naturalness.



Nor was the Explorer's good sound attributable to traditional "tube" sound. The highs were extended and clean—the jingling coins on Pink Floyd's "Money" (*Dark Side of the Moon*, Mobile Fidelity, UDCD 517) sounded like real coins. Similarly, the bass was solid, deep, and well defined: the bassoon and tympani on the "Peter and the Wolf" tracks sounded authentic.

The Explorer's sound remained uncongested and listenable even on heavy orchestral pieces such as the Brahms Symphony #4 (The Royal Philharmonic, Fritz Reiner, Chesky compact disk CD-6).

We don't mean to imply that the Explorer sounded 100% natural. Speech, including the narrator on "Peter and the Wolf," sounded slightly muffled; the applause on "Jerusalem" was a bit "off" (i.e., it sounded more like wind over the microphone than hands clapping). The Explorer lacks the extreme clarity and detail given by certain more expensive amplifiers.

The dynamics and transient response were very good when the fast-acting fuses were used. With slow-blow output fuses, these characteristics were still present, but the amplifier lacked the "startle" factor that is produced by better electronics. Similarly, the dynamic contrasts on the Corkhill percussion pieces were impressive with the replacement fuses installed. The individual drumbeats in the RSS demonstration were well-defined, fast, and solid.

At higher volumes some glare became apparent on midrange instruments such as horns. The Vivaldi trumpet concerto was sometimes difficult to listen to at high volume, although it was fine at our normal listening levels.

The Explorer was capable of reproducing plentiful ambiance information, particularly with the fast-acting fuses. On the recording of "Jerusalem," the space around and above the choir was clearly audible. Some ambient information, such as the applause on "When the Saints Go Marching In" (The Weavers, *Reunion at Carnegie Hall 1963*, Analogue Productions, APFCD 0005), was rendered slightly less impressive by the imaging uncertainties. Installation of the slow-blow fuses resulted in a small impairment of the ambiance recovery.

The Explorer produced stable, distinct images. The individual instruments on The Modern Jazz Quartet disk mentioned earlier were well defined; however, these images were always confined to the space between the speakers.

We never managed to solve the inconsistency problem. The sound seemed slightly more natural when the filaments were not continuously powered, but the effect was subtle enough that we could not draw a definitive conclusion.

**FINAL THOUGHTS**

**NM:** I enjoyed the natural, musical sound of the Kora Explorer 90SI. In fact, I thought the Explorer sounded better than other amplifiers I've auditioned which cost twice as much. I haven't heard anything yet in its price range with a more appealing sound.

If I had rated the Explorer while using the fast-acting fuses, the presence score in the sonic characteristics ratings would have been one point higher and the ambiance recovery score two points higher. Exchanging the fuses would not have changed the other sonic characteristics ratings.

**DM:** With the Explorer 90SI, I think that Kora has attempted to meld the best characteristics of vacuum tubes and transistors. They have succeeded in most areas. The bass is strong and well-defined, the highs are extended and clean, and the overall sound is natural and very listenable at reasonable volumes. At this price point, it's difficult to see how the sound could be much better.

I strongly suggest placing the Explorer on your short list of good prospects if you are considering an amplifier purchase in this price range (and if you exchange the output fuses). I enjoyed listening to it, but, as always, you should audition any amplifier in your own system before making a final decision.

with 1% THD occurring at 61W. The power available at 20Hz and 1% THD+N was 41W. At 20kHz I measured 45W into 8Ω at 1% THD. The maximum power available at 1kHz was 65W into 8Ω.

Switching to the A/V input, which bypasses the preamp tube and volume control, I saw 1% THD at 50W, with a maximum output of 63W at 3% THD. The clipping was symmetrical and quite rounded for a solid-state output stage (±32V). The designers have some flexibility in where the clipping occurs—tubes or FETs—with a hybrid design.

After these tests, the amplifier chassis was hot. The heatsinks, which do not come into contact with the chassis, were too hot to touch, as I found out while changing fuses. The fuse is located between each pair of heatsinks, so I needed to use a DIP IC removal tool to pull the fuse holder.

The distortion residual waveform for 10W into 8Ω at 1kHz (line input) is shown in *Fig. 2*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. This distortion residual signal is an asymmetrical second-order harmonic, with no evidence of any noise or fuzz. The distortion residual waveform for the A/V input at the same point is shown in *Fig. 3*, where the second harmonic has better symmetry.

The spectrum of a 50Hz sine wave at 10W into 8Ω, again with the line input, is shown in *Fig. 4*, from zero to 1.3kHz. The THD+N measured 0.74%, with the second, third, fourth, and fifth measuring −42dB, −79dB, −60dB, and −75dB, respectively. Higher-order harmonics are also prominent.

There are also power-supply harmonics at 120Hz (−85dB) and 180Hz (−77dB). I can't explain the two spikes at 70Hz and 170Hz, since they are not related to the input signal or the 60Hz power line, and disappear from the spectrum analyzer display when the input signal is removed.

The 50Hz sine-wave spectrum using the A/V input is shown in *Fig. 5*, where THD+N was lower at 0.54%. The second, third, fourth, and fifth here measure −46dB, −72dB, −59dB, and −73dB, respectively. The second has decreased, the third has increased, and the power-line-related harmonics are a bit lower. Those curious 70Hz and 170Hz spikes are identical to those of the line-input analysis.

*Figure 6* shows the Explorer's output spectrum reproducing a 12V p-p combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal into 8Ω, using the line input. The 1kHz IMD product is a high 0.28%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a 1kHz product of 0.075%. I repeated the test using the A/V input (not shown here), and the IMD was reduced to 0.021% and 0.041%, respectively. This would point toward preamp stage nonlinearity as the source of the intermodulation.

The 2.5V p-p square wave into 8Ω at 40Hz showed a fair amount of tilt using the line input, while it was much better with the A/V input. With either input, the 1kHz square wave was excellent, and the 10kHz square wave showed a slight rounding on the leading edge. When I connected 2μF in parallel with the 8Ω load (*Fig. 7*), the leading edge of the square-wave response showed some peaking, which is consistent with the extended HF response with this load.

*Manufacturer's Response:*

*Thank you for the review of the Explorer 90SI. We are happy to let you know that we have anticipated some of the remarks raised in the review with the following improvements:*

- *The fuses have been replaced by fast blow types as you recommended.*
- *The carton has been reinforced and foam added for better protection during transport.*
- *The front plate is now 8mm thick.*
- *The PCB has been reviewed to ensure a direct welding of the radiators in order to increase mechanical resistance and to protect the pins of the transistors.*

*Me, I like to see the tube behind the window! Don't you? Thanks again.* ❖

*François Philibert*
*Designer*

# Product Review

## Behringer Eurorack Mixers

### Reviewed by Gary Galo

*Behringer Eurorack MX802A and MX2004A Mixers. Behringer USA, 144 Railroad Ave., Suite 210, Edmonds, WA 98020, 425-672-0816, FAX 425-673-7647, e-mail: sales@behringer.com, www.behringer.de/eng/. MX802A: $179; MX2004A: $379. Warranty: One year parts and labor.*

Behringer International GmbH is probably unfamiliar to most readers. Based in Germany, Behringer specializes in affordable professional audio equipment, including digital effects processors, loudspeakers, microphones, compressors, and limiters. They also manufacture a line of vacuum-tube audio accessories, including a mike and line preamp, a parametric equalizer, and an eight-channel tube interface, which allows you to add a user-controlled amount of "tube sound" to the signal path.

They have rapidly become best-known for their Eurorack line of mixing consoles. Behringer has clearly targeted the Eurorack series to directly compete with the VLZ-series from Mackie Designs, an American pro-audio manufacturer based in the state of Washington. The Eurorack mixers are designed in Germany but manufactured in China. Behringer products were originally distributed in the US by Samson Technologies, a New York-based company. About a year ago, Behringer set up their own US distribution in Edmonds, Wash., in the backyard of their principal American competitor.

### MX802A

There are seven mixers in the Eurorack line, ranging from six to 32 inputs. In terms of complexity, the two models reviewed here represent the middle and lower end of the product line. The compact MX802A is billed as an eight-input, stereo out mixer (*Photo 1*). Four input channels, 1 through 4, are mono, and will accept both balanced mike (XLR) and line (¼″ phone) inputs. Standard +48V phantom powering is available at each microphone input. The line inputs will accept balanced (tip-ring-sleeve) or unbalanced (tip-sleeve) inputs.


**PHOTO 1: Behringer MX802A mixer. The compact design still offers a wide array of features and excellent performance. All audio input and output connectors are on the mixer top plate, as shown.**

Each mono input has a gain trimpot, a switchable 75Hz, 18dB/octave low-cut (i.e., high-pass) filter, a three-band equalizer, two auxiliary sends (pre- and post-fader), a pan control, and a level control. To make the mixer as compact as possible, all level controls are rotary pots, rather than slider-type faders. A peak light is included in each of the mono input channels to monitor mike preamp overload.

The stereo inputs are line-level only, and arranged in left-right pairs, channels 5–6 and 7–8. Each stereo input accepts two ¼″ mono phone plugs, balanced or unbalanced. Input level for each stereo pair is controlled by one pot, with a pan pot functioning as a left-right balance control.

A bit of logic has been built into the stereo inputs. If you connect a mono input signal to the left input, it is also routed to the right. In other words, if you connect an input to channel 5 only, this input is also routed to channel 6. The three-band equalizers and two auxiliary sends are also included on the stereo inputs.

The two auxiliary send buses feed unbalanced outputs. Two pairs of left and right balanced auxiliary returns are included, with the return level of each pair controlled by a separate pot. The main left and right outputs of the MX802A are balanced, and controlled by the stereo main mix level pot. A separate pair of control-room outputs are unbalanced, with level controlled by the same pot used to control the volume at the stereo headphone jack.

The MX802A also includes RCA tape-out and tape-in jacks. The tape outputs are in parallel with the positive legs of the main left and right outputs. You can switch the tape inputs to the main mix, or to the control-room monitor. Left and right output metering is accomplished with two rows of 12 LEDs. Green LEDs indicate levels of –20 to –6dB, yellow for +2 to +10dB, and red for clipping.

### MX2004A

The MX2004A is similar in conception to the MX802A, but with more inputs and greater flexibility. The MX2004A is billed as a 20-channel mixer, and if you

**PHOTO 2: Behringer MX2004A mixer. Up to 20 inputs, and considerable flexibility are offered in this fine performer. The logarithmic input and output faders feature a 60mm travel.**

include the two pairs of stereo auxiliary returns, it indeed is. Eight mono input channels feature XLR balanced mike and ¼″ phone balanced line inputs. Switchable +48V phantom powering is available on all eight mike inputs. Input gain trimpots and an 18dB/octave low-cut filter are included, just like the MX802A. The four unbalanced stereo input channels are line-level only. All input level controls are logarithmic slider-type faders, with 60mm travel.

The three-band input channel equalizers are enhanced to include a center frequency adjustment on the mid control—1kHz to 5kHz. Auxiliary send #2 is wired post fader, but #1 is switchable, pre or post. A solo button on each input channel allows that input to preempt all others at the monitored output, though these buttons never affect the main outputs.

A red solo light in the metering panel gives a visual indication that one of the solo buttons has been engaged. Each input channel also has a mute button, which also routes the input signal to the Alt 3 and 4 outputs. On the stereo inputs, the pan control functions as a left-right balance control, like the MX802A.

Each input channel is equipped with unbalanced channel inserts, for connection of outboard effects devices. Electrically, these insert points are located between the input preamplifier and the equalizer circuitry. You can switch the MX2004A's main balanced XLR outputs for line or mike level. A resistive attenuator is switched in to drop the output to mike level, if selected. A second set of unbalanced ¼″ main outputs operate at line-level only. The Alt 3 and 4 output pair are unbalanced and line-level.

The auxiliary send and return connections, tape in and out, and the control room outputs are configured like the MX802A. The output control section of the MX2004A features separate left and right output faders, plus a stereo ganged fader for the Alt 3 and 4 outputs. Control of the auxiliary sends and returns is, again, like the MX802A, with one important difference: you can route the Auxiliary 2 signal either to the main mix or to the cue feed. The MX2004A uses 13 rectangular LEDs per channel for output level indication, with the same color scheme as the MX802A.

## CIRCUITRY

Behringer does not include a schematic diagram with its products, only a block diagram. Specific circuit details are not possible, but you can make a few observations. Both the MX802A and MX2004A mixers come with outboard power transformers. These are not wall-warts! The transformers have both primary (120V AC) and secondary cords, which allows for more flexible placement. Six-foot cables on the outputs of the supplies ensure that the transformer hum field is kept far away from the mixer circuitry. The DC power connector is a screw-on, three-pin type, similar to the mike connectors used on CB and other communications equipment. Standard three-terminal IC regulators are in the mixer chassis.

Behringer prides itself on the extremely quiet performance of its mixers. Their ultra-low noise microphone preamps feature discrete input stages with matched, high-current transistors. All mixers in the Eurorack line incorporate the same mike preamp circuitry as their flagship Eurodesk 9000, which retails for $2600. In addition to their low noise levels, these mike preamps have an ultra-wide bandwidth of 100kHz.

Behringer engineers have also been especially careful in their selection of the IC op amp. The NJR NJM4580 is used throughout the Eurorack series. The 4580 was designed for low noise and low audio distortion. This dual op amp features a slew rate of 5V/μs, a gain bandwidth product of 15kHz, and harmonic distortion of 0.0005%. Input voltage noise is specified at 0.8μV RMS. The 4580 is the highest-performance dual op amp I've seen in an affordable mixer.

## PERFORMANCE

*Tables 1* and *2* list the manufacturer's specifications for the MX802A and MX2004A mixers. I was able to verify the Behringer claims for these mixers. I did note a slight peculiarity in the

### TABLE 1
### MANUFACTURER'S SPECIFICATIONS—MX802A

**MONO INPUTS**

Mike input: Electrically balanced, discrete input configuration
Bandwidth: 10Hz–60kHz ±3dB
THD+N: 0.007% at +4dBu, 1kHz
Mike E.I.N. (22Hz–22kHz): −129.5dBu, 150 source
−130dBu, input shorted
Gain range: +10dB to +60dB (mike); +10dBu to −40dBu (line)
Equalization: 12kHz ±15dB
2.5kHz ±15dB
80Hz ±15dB

**STEREO INPUTS**

Bandwidth: 10Hz–55kHz ±3dB
THD+N: 0.007% at ±4dBu, 1kHz
Equalization: Same as mono inputs

**MAIN MIX SECTION**

Maximum output: +22dBu balanced
Auxiliary send maximum output: +22dBm unbalanced
Control-room output: +22dBm unbalanced
Signal-to-noise ratio: 112dB, all channels at unity gain

**PHYSICAL**

Dimensions: 20/25mm × 160mm × 210mm
Net weight: 3.0kG (power supply not included)
Gross weight: 4.6kG

THD+N measurements for both mixers.

Below an output level of 0dB (reference to the metering on both mixers), the distortion is dominated entirely by noise. Beginning at an output level around 0dB, a slight nonlinearity shows up in the distortion waveform. This nonlinearity looks like a very slight power-supply instability. The THD+N still remains within the manufacturer's claim, however.

I have had an opportunity to use both mixers in recording and sound reinforcement applications at The Crane School of Music, SUNY at Potsdam, and have been extremely pleased with their performance. I normally used a high-end, two-channel microphone mixer for my live concert recording—a Sontec MB-1 that retailed for about $1000 when we purchased it about ten years ago. I certainly can't say that the Behringers are the sonic equal of my thousand-dollar stereo mike preamp, but I am amazed at just how good they really are.

### TABLE 2
### MANUFACTURER'S SPECIFICATIONS—MX2004A

**INPUT CHANNELS**

Mike input: Electrically balanced, discrete input configuration

Bandwidth: 10Hz–60kHz ±3dB

THD+N: 0.007% at +4dBu, 1kHz

Mike E.I.N. (22Hz–22kHz): −129.5dBu, 150 source −132dBu, input shorted

Gain range: +10dB to +60dB (Mike); +10dBu to −40dBu (line)

Max input: +12dBu (Mike); +22dBu (line)

Channel fader range: +10dB to −85dB

**EQUALIZATION**

High shelving: 12kHz ±15dB, "Q" fixed at two octaves

Mid Bell: 100Hz–8kHz ±15dB, "Q" fixed at one octave

Low shelving: 80Hz ±15dB, "Q" fixed at two octaves

Low-cut filter: −3dB at 75Hz, 18dB/octave

**MAIN MIX SECTION**

Bus noise, fader 0dB, channels muted: −100.0dBr (ref. +4dBu)

Bus noise, fader 0dB, all input channels assigned and set to unity gain: −88.5dBr (ref. +4dBu)

Maximum output: +28dBu balanced; +22dB unbalanced

Auxiliary send maximum output: +22dBm unbalanced

**GENERAL**

THD+N: 0.007% at +4dBu, 1kHz; below 0.02%, 22Hz–22kHz, normal operating levels, any input to any output.

Frequency response: 10Hz–120kHz, ±3dB, any input to any output

**PHYSICAL**

Dimensions: 40/90mm × 410mm × 385mm

Net weight: 6.0kG (power supply not included)


**PHOTO 3: Behringer MX2004A rear panel. The XLR main outputs can be switched from line to mike level. Channel inserts for the eight mono inputs, ¼″ main outputs, and the Alt 3 and 4 outputs are on the rear panel.**

What is especially impressive is the subjective absence of noise. These are among the quietest mixers I have ever used. The sound is also clean and detailed, and largely free of harshness and other unpleasant side effects.

Both of these Behringer mixers excel in sound-reinforcement situations, and if you need more inputs and even greater flexibility, several other more complex mixers in the Eurorack line should fit the bill. I am particularly enamored with the little MX802A, because so much has been packed into such a small space. For simple sound-reinforcement jobs—where only a couple of mike and line sources are needed—it is a great little mixer. It is also handy as a sub-mixer, augmenting your main board if you run out of inputs. And, its surprisingly clean sound will make it very attractive for location recording, especially where portability is an issue. Even at the suggested retail price of $179, it is a great value. At the street price of around $130, it is an unbelievable steal.

Some time ago Mackie upgraded its VLZ-series to the VLZ-Pro line, which features its XDR® (Extended Dynamic Range) microphone preamps. These preamps offer noise performance and dynamic range comparable to the Behringer mike preamps. At Crane we have a Mackie 1402-VLZ Pro mixer in the recording booth of our small lecture/recital hall. The 1402 is comparable in inputs and features to the Behringer MX2004A, but with only six microphone inputs.

So far, I have a slight preference for the Behringer in sound quality. Both mixers are noise-free, but the Behringer seems just a bit more transparent. This may be due to Behringer's use of the superior 4580 op amp. Mackie uses the 4560, which is a bit slower (4V/μs versus 5V/μs) and narrower in gain-bandwidth product (10MHz versus 15MHz). The 4560 also has lower output current and a slightly higher noise level than the 4580.

Price also tips the balance in Behringer's favor. The US-built Mackie retails for $629, and the typical street price is around $435. The Chinese-made Behringer retails for $379, and usually sells for around $275. I have spoken with other users, and two pro-audio dealers, and can't find a clear consensus regarding a sonic preference for the Mackie versus the Behringer products. Both firms seem to have enthusiastic followers.

Behringer's English instruction manuals are very clear and well-written. But, Mackie definitely wins the prize when it comes to user-friendly, nonintimidating manuals. The Mackie manuals include an abundance of connection diagrams. Construction quality on the Mackie products is excellent, perhaps a bit more robust than the Behringer mixers. But, the two Behringer samples under consideration here are still very well-made.

I did have one problem with the MX2004A—the channel 2 input fader was intermittent. Behringer referred me to a factory-authorized service center in New York City. The input fader was replaced, with a total turnaround time of about two weeks, including shipping, which I consider very reasonable.

Behringer has made two very impressive contributions to the low-priced mixer war with the MX802A and the MX2004A. These products offer amazing performance and flexibility for the money, and are highly recommended. ❖

# Xpress Mail

## DIPOLE SPEAKERS

I am interested in building the speaker system described by Tom Perazella ("On Angel's Wings," Jan. and Feb. '01, *aX*). I have researched this type of system and have found no information. Since this is an unusual system, I would appreciate a theory of operation explanation. I am not sure which speaker is left or right or how to incorporate a midbass (70–200Hz) system for some home-theater punch!

Pat Freeman
Suttons Bay, MI

*Tom Perazella responds:*

*Two good sources of information about planar dipole planars, especially the BG RD75, are the websites of BG Corp. and John Whittaker. The URLs are:*

*www.bgcorp.com*
*www.csulb.edu/~jwhittak*

*The BG site contains an Acrobat file of a paper by Keith Yates that gives practical insight into the differences between the more common point source and the line source, of which the BG RD75 is one example.*

*Other information available on the BG site includes product sheets giving specifications of the whole RD series of drivers. Newer drivers using neodymium magnets are also discussed.*

*The site from John Whittaker is quite extensive. Mr. Whittaker and Rudi Blondia did quite a bit of research into planars. Much of this research was done in a large gymnasium and outdoors. In addition to their research, there is an extensive list of references if you would like to do further research.*

*The completed speaker that is displayed in my article is the right-side version. A mirror image is used for the left. Since there is only a small baffle on the left side of the right-side speaker, there is considerable cancellation on that side from the back wave. This helps widen the image. The corresponding holds true for the left speaker.*

*For midbass—from 60Hz to 250Hz—I am currently using two 12″ HSU woofers in sealed boxes. The performance is good, but the radiation pattern, as you can imagine, does not match the RD75s in their baffles. My plan is to build two line-array dipole woofers to cover that range. The design is not final yet, but I will use six Peerless 831727 10″ woofers on each side. Since I don't like to use series arrangements of drivers, it will use three Audiosource Amp Three stereo amplifiers. Each will power two paralleled pairs of the Peerless woofers.*

*Frequency steering will be done by my custom active crossover. The woofers and amps have already been purchased. That should force me to complete the design in the near future. When done, I plan to submit the project to* audioXpress.

*If you plan to use a dipole woofer configuration, understand that you will need a lot of volume displacement capability from the drivers. Depending on the design, room, and crossover points, you may also need frequency contouring. The 831727 has an $X_{MAX}$ of 9mm. Even with that level of excursion, I will still need six per side to provide the volume displacement needed.*

*I hope this information will provide some guidance for your project. Have fun!*

## PHONO PREAMP DESIGN

I thank Norm Thagard for his two-part phono preamp article (Jan. and Feb. '01). On all the stuffing guides, J4/J6 are inverted. I have replaced Q1/Q4 with 2SC 3381/2SA 1349, which works well. I would like suggestions to use it as a linear preamp.

Carlo Ghersi
Genoa, Italy

*Norman Thagard responds:*

*Mr. Ghersi is correct; J4/J6 are transposed on the stuffing guide. It is important to note this, since J4 is a dual p-channel device, while J6 is a dual n-channel device.*

*I am sure that most small-signal BJTs would work in the Q1-Q4 positions. The advantage of matched devices is that generation of the desired current ratio and constancy of that ratio is better. The second stage of the preamp is a linear stage. If no more than 40dB (100) of closed-loop gain is needed, that stage, itself, can serve the purpose.*

*If you require greater gain, then eliminate C2 and R3 from the first stage feedback network. Set the value of R4 so that your desired stage gain is achieved when R4 = (gain – 1)R2. Overall gain will be first stage gain times second stage gain and can range from unity up to 80dB (10,000) or so.*

*It is strongly recommended to increase R2 to 499Ω at a minimum, unless noise considerations are very important. This is because the low feedback network resistance will otherwise heavily load the first stage output. I was pleasantly surprised to find that the phono preamp performed very well despite the load imposed by the first stage's feedback network. It allowed me to make low-noise performance the principal design criterion. However, I usually avoid such heavy loading and would boost the values of R2 and R4 proportionally since it is their ratio, not their absolute values, which determine stage gain.*

*Since all this advice is given on the fly, I cannot take responsibility for the results of any alterations to the published circuit except for the obvious stuffing guide correction. Even so, I have experimented extensively with the basic single-stage op-amp configuration represented by the linear second stage of this preamp. It is my plan to construct a preamp based on this topology. The preamp will feature single or differential input and output capability. The published phono preamp circuit will be the phono section in this preamp.*

## HORNS AND MIDS

In his letter (*SB* 7/00, p. 38) about G.R. Koonce's article, Bill Fitzmaurice states that he tested a horn with two drivers. He found that the SPL increase was not linear and that it didn't seem to be the 6dB that Mr. Koonce said it would be, when comparing two drivers connected versus only one connected. Then he went on to make the assumption that, because a horn's output is largely determined by its mouth area, the SPL did not increase the promised additional 3dB because the mouth of the horn didn't change.

Well, what makes pressure is not moving an area of air but a volume of air. The fact that the mouth area does not increase does not mean that the pressure would not increase. Just remember that, as the drivers move further back and forth, the volume of air displaced increases and hence the SPL increases.

Professional audio uses what I know as manifolds (*Figs. 1* and *2*). These use multiple drivers that all fire into a single cavity. Such a cavity is then exposed to the area to be pressurized through what you could call a mouth (not necessarily a horn mouth, but a mouth nonetheless). Even if I were to retain these cavities' mouth areas, SPL will increase as I add drivers (with the same power input). Of course, the lower the frequency played, the more this is true.

The whole idea is to try to positively couple as much of the outputs of the drivers as possible. The effective change to the system is an increase in the volume of air displaced as opposed to an increase in area.

But what would happen if you were to simply unplug one of the drivers while keeping it mounted to the manifold? The manifold's cavity would now be energized by three drivers, and it would be tuned by a passive radiator and a port (the port being the area and depth of the mouth). This would not provide a sample for appropriate measurement purposes, since the manifold with the four drivers is only being tuned by the mouth (port). So, the only trustworthy way to trustfully measure the difference is by replacing the disconnected drivers with a solid element (as Mr. Koonce suggested).

Mr. Fitzmaurice's results do not surprise me, since I have tested many manifolds and have, in fact, tuned some of them to play "differently" by simply connecting and disconnecting drivers. What I can tell him for sure is that the more rigid the cavity's walls, the more drivers I use, the closer these drivers are together, and the higher their air-displacement ability, the higher SPL I will get. And my SPL gains seem to come in 3dB increments for every doubling. This set of conditions of course reach a point at which they no longer apply due to compression caused by losses from tremendous vibration, flexing, and the fact that the air between the drivers and the mouth begins to work much more like a spring. But these problems have nothing to do with there being one, two, or three drivers. They are due to some serious dBs!

I hope Bill Fitzmaurice will accept this as an observation rather than a criticism. I am all for learning things out of observation and experience. I have learned quite a bit through such methods.

Also, I have a suggestion for Jim Moriyasu. Has he considered the effects that the proximity of the different cavities' internal walls would have on the tests? At the beginning of his article (*SB* 7/00 and 8/00), he shows how even large openings right by the rear of a loudspeaker can cause negative system results. It just happens that while looking at his photos, it occurred to me that perhaps the cylindrical cavity would perform very poorly because it would be equivalent to a very small mounting hole on a very thick front baffle. Also notice that the two cavities that place their inner walls farthest

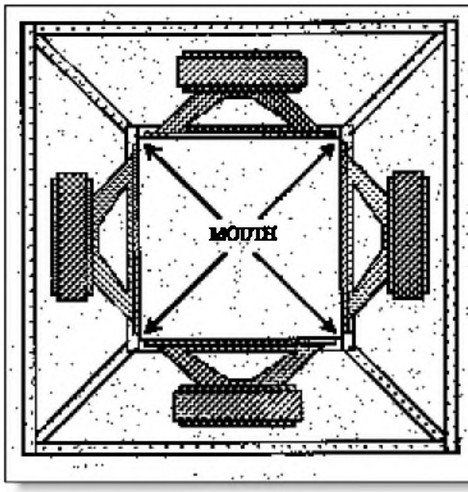



**FIGURE 1: Front view of sample manifold driver arrangement.**

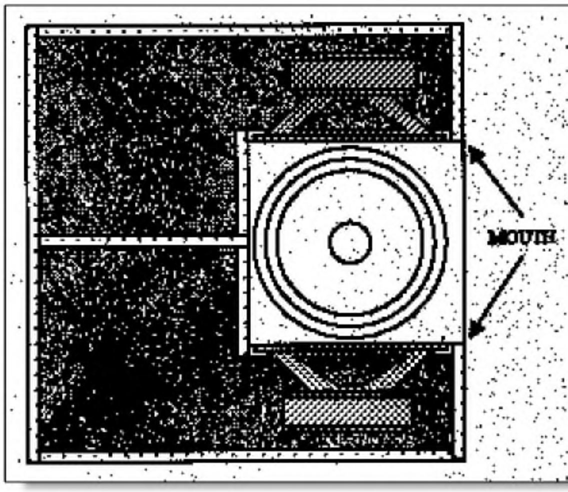



**FIGURE 2: Manifold side view.**

from the rear of the speaker—the rectangular one and the cube—were the two that did the best.

So, how about testing his speaker in enclosures of similar shapes but that take up a much larger volume? After all, most of the negative effects seem to be most visible at frequencies that should not be affected by a larger box.

Jim Moriyasu's experiment was very interesting, and I would like to congratulate him for being willing to share his findings with all of us. It must have taken a lot of time and effort to go through all of what was needed to get to the point of being able to write the article. Thanks!

Alberto Lopez
Phoenix, AZ

*Bill Fitzmaurice responds:*

*It seems to me that reader Lopez basically agrees with my conclusions. He has found that adding more drivers to a horn throat will give 3dB of added sensitivity for every doubling of the number of drivers, when the mouth area remains constant. This is precisely what I have found to be true.*

*As for the mass of the air column within the horn, I am currently of the opinion that increasing the air column mass will serve primarily to lower the horn's passband frequency, and that raising sensitivity is the product of a larger mouth opening.*

*As for the "manifold" driver arrangement, this is well known, being commercially available from ElectroVoice and other vendors. By using four drivers wired as series/parallel pairs, you can get a lot of power handling, making this a seemingly attractive proposition in high power/long-throw applications. I have turned my attention away from "manifold" designs. My latest horns are getting sensitivities so high that they don't require more than one driver for adequate power output.*

*Mr. Lopez is also correct in that if I had tested my Doppelganger with one driver electrically disconnected, but still mounted in the cabinet, the disconnected driver would have acted as a passive radiator within the horn throat. That would have rendered the experiment invalid. When I tested the cabinet with only one driver, the second driver had been removed from the cabinet, and its mounting hole on the baffle sealed.*

*Jim Moriyasu responds:*

*I'm always glad to share my findings and I look forward to hearing from others. Your intuition appears to be correct regarding larger midrange enclosure volumes as part 2 of the article attests. And I think you are correct in pointing out that the closeness of the internal walls, especially near the rear of the driver basket, will cause problems in the midrange's response. I think the smaller enclosures also perform poorly because the smaller enclosure volumes lead to higher pressures, especially with regard to enclosure-induced resonances. This is why damping material is unable to rectify the resonance problems and response aberrations.*

## WISH LIST

Love your magazine. First, I would like to see fewer reviews of already built components and more reviews of kits and speaker drivers.

Second, on the home construction front, could anybody out there write an article about chassis design? This may sound trivial, but it is what defeats me from a true DIY construction project,



leaving me the choice of upgrading an existing chassis or buying a kit. I think an article could address layout, wire selection, wire routing, and component selection. Insights on mounting switches, potentiometers, and so on, would also be helpful. It would also be a benefit if the writer assumed that the lowly reader does not have access to a fully equipped shop or state-of-the-art tools. Thanks for listening.

Richard Jenkins
jenkinr@napapipe.com

## REVIEW FEEDBACK

I'm writing this letter to express my disappointment at recent issues of *audioXpress*.

*audioXpress* has now reached issue number 4, which in total would contain the same amount of tube articles as one copy of *Glass Audio* would have, and I'm afraid it just doesn't compare. What I found particularly interesting in *GA* and *AE* were the articles on clever gadgets, which you could incorporate into an existing system and enhance them. Once you spend your money on a tube power amp—either bought or built—you have little need for another one in the near future. (And if there's another Satoru Kobayashi amplifier I'm going to scream!) I have little interest in any more speaker building (as with power amps) or digital electronics.

So looking at issue 4 I found little of interest, except the review of the Mapletree Octal 6. Charles Hansen's description and measurements are very thorough (as always—kudos here!) and generally substantiate the manufacturers' own data. However, measurements are somewhat meaningless (as has been pointed out in the past), so I looked forward to the listening report.

And here's my gripe (and the reason I write at all): how on earth could the MacArthurs have been allowed to proceed, with such a mismatch and inappropriate equipment? I think it highly irresponsible of *aX* to even have printed it, as it can be the kiss of death to a small manufacturer! Particularly as Eric Barbour in the very same issue is concerned with the viability of the tube industry and is looking at ways to reduce the high cost and effort to built tube gear. The manufacturer was very graceful in his reply, but personally I think a public apology and a new listening report (but probably not by the MacArthurs again) would be in order. Please be more thorough.

As far as I am concerned, I would have been happy with Charles doing the listening as well; he could verify measurements with the aural results. And one suggestion to the manufacturer: why not incorporate the CF output as permanent output and the other modes as input options? As the MacArthurs rightly pointed out, you wouldn't buy a preamp for its passive volume control.

I wish *aX* all the best.

Pierre Drion
pierredrion@optusnet.com.au

*Duncan and Nancy MacArthur respond:*

*As we stated in our critique, the maximum output voltage of our DAC is 2V. This is a standard output level among CD players and DACs. The input sensitivity of the Stingray is 185mV for full output. This is less sensitive than some power amplifiers that are currently on the market. The preamplifier in CD-based systems is most often operated with a gain of less than one—we just admitted it up front.*

*The Stingray was chosen for use in reviewing the Octal 6 because it was the most transparent-sounding amplifier that we had available at the time. Although a consumer would probably not use these components together, we see no specification that would imply that this is an incompatible combination.*

*We are particularly concerned about any implied correlation between the size of the manufacturer and the quality of the sound. Although we made a major effort with this review to phrase negative assessments tactfully, we also have an obligation to report honestly on what we've heard. We believe that we should report on the sound of a unit irrespective of the size of the manufacturer or the type of circuitry used.*

*We also question the term "verify measurements with the aural results." If the aural results always "verified" the measurements, then there wouldn't be much point doing listening tests. In our listening tests of the Octal 6, we heard nothing that particularly disagreed with the measurements, but a good set of measurements does not necessarily imply good sound.*

## 5687 VS 6SN7

Regarding the discussion on "Driver Circuit for 6AS7 Amp" (Xpress Mail, *aX* Jan. '01, p. 101), I cannot reply to the validity of Mr. Wong's linearity claims for the 5687 either, as I haven't used them in any of my circuitry, although I have some in stock. I also prefer octal tubes over miniatures. The main differences between the tubes are that the 5687 has a lower plate Rp (3000 vs. 7700), higher Gm (5400 vs. 2600), and higher plate current for 250V (12mA vs. 9mA). The 6SN7GTA/B has higher plate ratings (5.0W vs. 4.2) and higher gain (20 vs. 16), while total plate dissipation for both tubes is 7.5W. Both tube types are capable of high current peaks. The 6SN7GTA/B is capable of operating at the same power/current rating as the 5687. With that in mind, there may be little if any difference between the tubes.

Remember that a comparison (in this case at least, since both can be operated at the same points) must be done under the same operating conditions to draw any valid comparisons. Most tubes will perform better with heavier current draw, i.e., more linear. Another advantage of the 6SN7GTA/B is that it is rated to 450V plate vs. 300V for the 5687. (Even given a 10% increase for the different rating system, the 5687 is still well below the 6SN7.) Mr. Wong's circuit is operating on the edge of maximum plate voltage ratings. Additionally, both tube types were characterized for vertical output operation with high *reverse* voltage ratings, which has no effect on amplifier ratings.

I also agree with Mr. Stewart and Mr. Cottrell that Mr. Wong's driver circuit is a very capable high output driver despite the weakness in the input circuit; the differential amp is an excellent circuit that deserves much more use in audio. It was used to considerable benefit in instrumentation and analog computers, to name a few.

On a side note, I'm pleased to see that you've kept reasonable balance among the types of articles in your magazine, unlike others who have consolidated and completely changed the character of their publications.

Edwin G. Pettis
Grand Junction, Colo.

*John L. Stewart responds:*

*Thank you for the opportunity to respond to reader concerns and comments. As always, there are trade-offs, and you must decide where to put your efforts and resources.*

*Having said all that, I had for the most part put the "A Different Triode Power Amp" (GA 2/99) out of mind. Instead, I charged ahead with some new projects which I hope you will enjcy in this magazine soon.*

*The 5687 was originally developed for industrial and military applications. Probably audio was not in mind, let alone high fidelity. Let me briefly bore you with some history of my own experience.*

*Before the advent of low-cost semiconductor devices, there was lots of research into better kinds of magnetic memories. One*

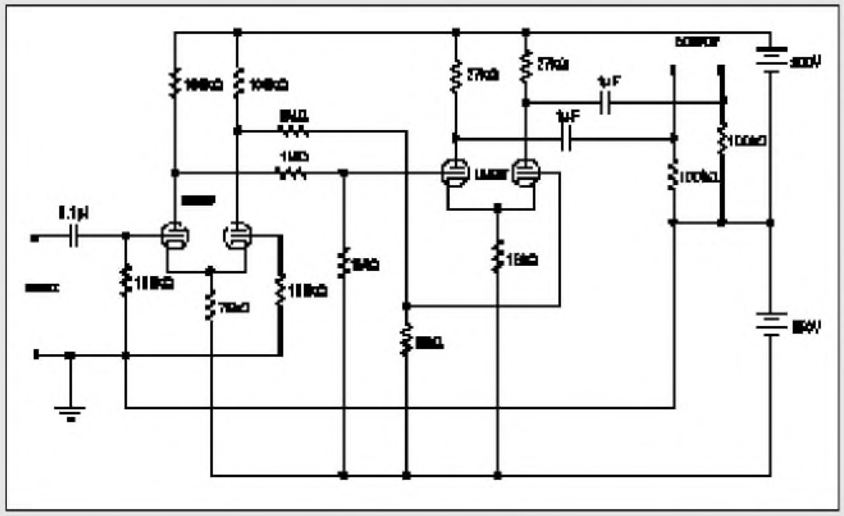


FIGURE 1: DC-coupled differential amplifier.

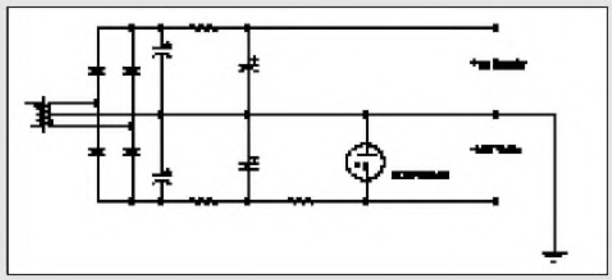


FIGURE 2: Differential amp power supply.



*of the projects I took part in was an investigation into the possibility of a magnetic core memory system where the contents were non-destructive. That doesn't mean it was liable to expire, but rather that the contents of the memory (zeroes or ones) were erased each time the memory was interrogated. Then the information would need to be replaced after each memory cycle by another circuit. That used up some more time and was something else to fail.*

*Our team needed some very fast electronics to generate the pulses used in this investigation. The 5687, having a relatively high perveance, was a good choice for some of this work. Another double triode equally suited for this application was the 7119. We used lots of them. It never occurred to me at the time to try them in an audio amplifier.*

*Other tubes we used included the TV Horizontal Sweep Tube 6DQ5, which was capable of pushing a 1.5A pulse into a 50Ω transmission line with a rise time of about 40ns. I once built a 100W amplifier using four of the closely related 6CD6GA, which used 980V on the plates. A really fast tube is the Secondary Emission Pentode EFP60. It managed up to 200mA pulses with a rise time of about 3ns.*

*We made extensive use of differential amps. Some of them were commercially available from Philbrick, Burr-Brown, and others. We had in the lab four independent power supplies, all wired into three separate workbenches so that many circuit configurations were possible.*

*Voltages available were ±300 and ±150, all regulated. Keep in mind this was all done around 1960. We needed to devise ways of looking at this high-speed phenomenon as well, but that is another story.*

*Now that we are in a vacuum-tube revival in audio, where do we get something to work with? What has not been smashed is showing up as surplus or NOS (New Old Stock) and everybody wants some. That includes tubes and other equipment originally not meant for audio.*

*I agree with Mr. Pettis that to make a valid comparison the various devices should be tested under similar conditions. However, for my critique of Mr. Wong's circuit, I needed an actual measurement of the 5687 zero bias line, from tubes currently available on the market. How often do we manage to acquire product that doesn't live up to the published specs?*

*Mr. Wong was good enough to e-mail me a copy of the complete 5687 data sheets. I was pleased to see that my four samples* were close to the published specs. With this information I was able to predict a 28% increase in the available voltage swing when using the 5687 as opposed to the 6SN7 family.

A look at the plate family of curves told me the distortion at all levels should be relatively low, at least competitive with the 6SN7 family. You would need to build similar amps with these tubes and test them under similar conditions to get the real answer. No doubt the 5687 would be the winner at high levels, where the 6SN7 family can't go.

Use of both these tubes as vertical amplifiers strikes me as a convenience for the TV industry until suitable tubes were developed, and there are many. I used one (6EA7/6EM7) in the article "An Affordable SE Triode Amp" (GA 4/00). Several others would have worked as well.

The differential amplifier is a great way to go when driving any push-pull circuit because of its inherent balance. Although industry used the diff amp extensively, they did not appear often in the audio world. That was probably due to the need for a second high-voltage supply.

This would often be negative, which wasn't convenient in the days before silicon rectifiers. Mr. Wong, by using a semiconductor to sink the current in his first stage, has avoided the need for a negative HV supply. My only comment is that the 9V supply does not leave much room for the circuit to move—18V would be better.

The second stage is direct-coupled to the first, which is an advantage. In this circuit the 5687 sees only 170V DC (300-130) at the quiescent point, which is OK. However, because of the way in which the DC coupling is accomplished, it needs a much higher supply voltage, which I would prefer to avoid. Another problem shows up in the heater-cathode insulation rating of the 5687: only 90V. The circuit as it stands may need another heater winding.

Level shifting in the coupling system between stages one and two provides an easy way out. It uses a negative supply to sink both cathodes as well (Fig. 1).

A simulation by Electronic Work Bench software showed this circuit to have a gain of more than 400 from the input to each of its outputs. I used the 12AX7/12AT7 combo since they were available in the simulator software. It still doesn't tell us anything about linearity, but I think the 5687 would be better than the 12AT7 in this circuit as well.

An important advantage of this configuration is that the required negative supply can easily be produced by the same winding on the power transformer as that which provides the positive supply (Fig. 2). I usually limit the negative supply to 150V in order to protect heater-cathode insulation during the warm-up period. I like a one-tube solution, such as the OD3/VR150 Gas Regulator. The rest of the circuit constants will depend on what I am doing in the amplifier.

Another solution to the problem of driving an output stage was suggested by George David (GA 5/99). I was aware of this circuit, since it appears in my copy of the RCA Receiving Tube Manual RC16 from July 1953. George was good enough to send me a copy of the original article describing the complete amplifier. It uses one of the 6SN7 family, as well.

In summary, all of these amplifier driver stages can benefit from the simple circuit rearrangement I suggested in the article covering the Different Triode Power Amp. Typically you can expect to double the output voltage capability of your driver amp. I hope you will try it. ❖

# New Chips on the Block

## Analog Devices AD1833 Multichannel Audio DAC

**By Charles Hansen**

The AD1833 is a complete high performance single-chip, multichannel, digital audio playback system. It features six audio playback channels, each comprising a high-performance digital interpolation filter and a continuous-time current-output analog DAC section. It uses the same design as the AD1855 stereo DAC. Other features include an on-chip, 1024-step linear (optional log) clickless attenuator and mute capability, per channel, programmed through an SPI-compatible serial control port.

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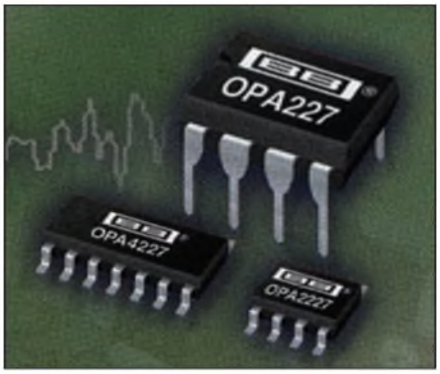
# New Chips on the Block

## Burr-Brown OPA227/228 Low Noise Op Amp

**By Charles Hansen**




**FEATURES**

Low noise: 3nV/√Hz.
Wide bandwidth:
OPA227: 8MHz, 2.3V/µs
OPA228: 33MHz, 10V/µs
Settling time: 5µs (significant improvement over OP-27).
High CMRR: 138dB.
High open-loop gain: 160dB.
Low input bias current: 10nA maximum.
Low offset voltage: 75µV maximum.
Wide supply range: ±2.5V to ±18V.
OPA227 replaces OP-27, LT1007, MAX427.
OPA228 replaces OP-37, LT1037, MAX437.

The OPA227 and OPA228 series op amps combine low noise, low distortion (0.00005%), and wide bandwidth with high precision to make them the ideal choice for applications requiring both AC and precision DC performance. The OPA227 is unity-gain stable and features high slew rate (2.3V/µs) and wide bandwidth (8MHz). The OPA228 is optimized for closed-loop gains of five or greater, and offers higher speed with a slew rate of 10V/µs and a bandwidth of 33MHz.

The OPA227 and OPA228 series op amps are ideal for professional audio equipment. In addition, low quiescent current and low cost make them ideal for portable applications requiring high precision. They are pin-for-pin replacements for the industry standard OP-27 and OP-37, with substantial improvements across the board. Dual and quad versions are available for space-savings and per-channel cost reduction:

| Single | Dual | Quad |
|---|---|---|
| OPA227 | OPA2227 | OPA4227 |
| OPA228 | OPA2228 | OPA4228 |

The OPA227, OPA228, OPA2227, and OPA2228 are available in DIP-8 and SO-8 packages. The OPA4227 and OPA4228 are available in DIP-14 and SO-14 packages with standard pin configurations. Operation is specified from −40°C to +85°C.

### BUDGETARY PRICING

Pricing will vary according to grade and package; 1,000 piece; recommended resale in US$; FOB USA:

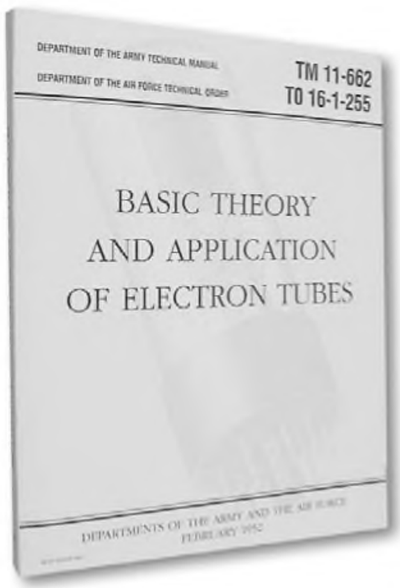
OPA227: $1
OPA2227: $1.70
OPA4227: $3.80
OPA228: $1
OPA2228: $1.70
OPA4228: $3.80

Texas Instruments, Burr-Brown Products, PO Box 11400, Tucson, AZ 85734, 1-800-548-6132, www.burr-brown.com. ❖

# Book Review

## Basic Theory and Application of Electron Tubes

**Reviewed by Larry Lisle**




Quick, name four methods of electron emission from the surface of a metal! If you had read Chapter 2 of *Basic Theory and Application of Electron Tubes*, the latest reprint of a golden age classic, you would be able to rattle off the answer. (Available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, (603) 924-9464, FAX (603) 924-9467, e-mail custserv@audioXpress.com, $16.95, 215 pages, with index.)

The title pinpoints what the book is about. The topic has been covered in many other texts throughout the tube era, but what's outstanding about this book is how well it's written.

The book began life in 1952 as an Army and Air Force technical manual. The people involved had plenty of experience teaching newcomers to electronics about tubes, in the least amount of time and with the best results. The sentences are unambiguous declarative, each intended to transfer a bit of knowledge.

Parts where students had problems with misconceptions in the past have been anticipated and explained in detail. The illustrations are superb. This book was written by people who could teach!

### CHAPTER SUMMARIES

The book is divided into 12 chapters. Chapter 1 begins with a brief description of how electron tubes have been used and a short history of the Edison effect, the Fleming valve, and the DeForest audion, or triode. The chapter ends with an outline of tube functions such as rectification, amplification, and oscillation, which will be covered in depth later.

Each chapter has a summary and review question. These are very helpful to a student using the book for self-instruction.

Chapter 2 covers electron emission in a very clear fashion. I wish I had read this chapter before trying to tackle some of the classic papers! It describes types of emitters, the materials commonly used to make them, and their advantages and disadvantages.

Chapter 3 is titled "Diodes." Don't be tempted to skip this chapter, because it lays the foundation for the fundamentals of vacuum tubes. Some of the topics covered are basic tube construction and schematic symbols, the behavior of electrons in an electrostatic field, the space charge (very well done), vacuum tube characteristics, characteristic curves as cause and effect, linear and nonlinear characteristics of tubes, families of curves, and on and on. These basic concepts are wisely introduced with the study of two-element tubes.

Again, don't skip this section. With home study (about the only way to learn about tubes these days) it's so tempting to go on to amplifiers. But if you do, your progress will be limited to copying existing circuitry because you won't have a real understanding of the theory on which the circuits are based.

Chapter 4 is titled "Triodes." It begins with the physical construction of triodes, which includes diagrams showing various grid shapes and a list of metals often used for grids. How a triode works is covered next. This part occupies 43 pages of the text and is not something you're going to absorb in a single evening! Rest assured, though, that if you take the material in small doses and review as needed, you will understand how triodes work.

Chapter 5 covers multi-element tubes. Depending on your interest, you may or may not wish to cover this chapter, but you *will* want to read Chapter 6, "Amplification," and Chapter 7, "Amplifier Gain and Coupling."

Chapter 8, "Rectifiers and Detectors," is useful if you're planning to use tubes in the power supply and helpful in understanding solid-state circuits as well.

### WRAP-UP

The other chapters are handy to refer to, although they might not have the wide appeal of those I've listed. Items of special interest include the sections on transmitting tubes and electron-ray indicators.

This is an excellent book, delivering what the title says it will better than any other currently published text I've seen. Its only rival is the out-of-print Navy version. The text does assume that you are familiar with basic electronic units such as the volt and with basic components such as resistors.

It also doesn't present circuits of complete amplifiers, so if you have no previous knowledge of electronics, you'll need to supplement this book with others from the same source or elsewhere. But you won't need to look elsewhere for material on how tubes work! ❖

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# Yard Sale

I have a number of back issues of *Glass Audio* that I must get rid of. I hate to throw them away because some audiophile might well treasure them. Will pay postage. Joseph Mann, jdm49506@aol.com.

## FOR SALE

SV 572-3 (QTY-2), new, $10 each; 6BQ6 or 6CU6, NOS, $10 for 5, minimum order, plus $5 shipping. Joseph N. Still, 306 M Canterbury Rd., Bel Air, MD 21014, 410-638-5996.

Philips CDC-875 changer with Crown DAC, mint, $190; Radio Shack headphone amp deluxe, $19; Elektor DAC kit (PCM-63), all parts, $199; Marantz 23 AM/FM tuner, mint, $149 ppd.; IR2200 imager kit, unbuilt, $79. Dave, 724-274-8149.

Two pairs Magnequest TFA-204 SE output transformers, 3K: 16, 8, 4Ω, 8W, 60mA, unused pair, $175; slightly used, leads not cut, $150/pair; Hammond 714 enclosed power transformers, 510-0-510V, 170-250mA, leads cut, $45/pair. Mark, 541-388-7584.

## Test Tracks

*from page 96*

sure). The result? Guaranteed to clear the room every time!

For the bold or foolhardy, it is music of obscenely frightening proportions. This is stuff with immense rilking potential (doubly so, if you purchase the LP). Needless to say, the sound is of reference quality. And please, for your own safety, keep the Holy Water nearby! ❖

John Suslov
Santa Rosa, Calif.

*Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to "Test Tracks," Audio Amateur, Inc., Box 876, Peterborough, NH 03458. We will pay a modest stipend to readers whose submissions are chosen for publication.*

# Test Tracks

## Reader-submitted selections to test audio systems.

**1. Grace Jones, *Private Life: The Compass Point Sessions*, Island Records Inc. 314 524 501 2 (314 524 502 2) (2 CD).** Elegant Parisian disco-chic! Grace is suave and sophisticated, world-weary, primal, and sexy. Those of you familiar with the movie *Frantic* will remember the song "I've Seen That Face Before" (track 8, disc 2). It is an especially evocative number, whose essence is equally well portrayed in the film. The sound quality is of near-reference quality (for a pop recording from the '80s).

Be prepared to have your chest cavity pummeled by Jah Wobble's bass and your senses scorned, abused, taunted, caressed, violated, and ultimately seduced by Grace's all-powerful vocal delivery. Long live the Queen!

**2. Arthur Honegger, *Symphonies no. 2 and no. 3 "Liturgique,"* Igor Stravinsky, *Concerto in D for String Orchestra*, Herbert von Karajan (cond.) Berliner Philharmoniker, Deutsche Grammaphon 447 435 2 (CD).** A newly remastered treasure from DG's LP catalog. Honegger alone is reason enough to purchase this disc. Can music be as direct as speech? In the case of Honegger, you can hear his thoughts on death, war, and the folly of human endeavor.

But this is not "program music"; it is deep, ruminative meditation. Challenging perhaps, but you will be the better for it, trust me. The sound: fabulous string tone, superb separation between instruments, and impressive tonal/dynamic contrasts. Puts to shame 99% of current "DDD" recordings.

**3. Tony Williams, *Lifetime,* Blue Note (RVG) 7243 4 99004 2 4 (CD).** An absolute classic! An exercise in restraint, subtlety, and texture. Sam Rivers at his haunting best. Tony Williams doesn't just keep time, he *makes* time! Nobody plays bass like Ron Carter (except maybe Gary Peacock, who's also on this album). Lovers of bowed bass, you need this! Although some may classify this as "free jazz" (and therefore inaccessible), you may casually ignore such naysayers. This is beautiful, wistful, dark, and mystical music. Take track 5, "Barb's Song to the Wizard" (with Herbie Hancock), as a perfect example of this. Sonically, this is one of the best of the recent RVG remasters, which means that the sound is both richer and fuller.

**4. Du Caurroy, *Requiem Des Rois De France*, Ensemble Doulce Memoire, Astree Auvidis LC 7496 P1999 (CD).** A rare treat! Somewhat obscure, but it is in print and worth the search. A requiem for the French monarchs of the *Ancien Regime*—in full Counter-Reformation splendor!

The sound (and performance) are impeccable. The diction, intonation, and control are flawless. The portrayal of ambient/acoustic space is beyond impressive. Beware, however. Some of the complex harmonies can reveal intermodulation nasties hitherto unnoticed, so make sure the rest of your system is up to the challenge.

**5. Stan Getz, *Plays Music from the Soundtrack of the Motion Picture "Mickey One"*, Verve, SE4312 314 531 232 2 (CD).** Stan Getz and Eddie Sauter do Xtreme Continental, but with plenty of substance and emotion. In fact, despite its "spaciousness," this music can sound downright claustrophobic and depressing (and that's a good thing).

Stan Getz apparently felt a kinship to the tragic persona of Mickey One, because this music is deeply felt and personal. Although it may initially be a crowd-pleaser, its introverted soul is quickly apparent. The Verve remasters are among the best of the recent A to D transfers. The clarity and dynamics are stunning, while managing to retain a fair bit of analog—like warmth and tonal purity. The result is very impressive indeed. This is a soundtrack for lost souls; i.e., do not consume with alcohol!

**6. Charles Alkan, *Piano Music of Charles Alkan*, Liszt, *Hexameron*, Raymond Lewenthal (forte-piano) BMG 09026 63310 2 (CD).** Before there was Marc Andre Hamelin, there was Raymond Lewenthal—His Royal Byronic Majesty. Hamelin has yet to be photographed in a cape! Lewenthal and his image (not to mention pianistic sorcery) are the perfect vehicles for this music (read the liner notes for Lewenthal's thoughts on "rilking").

Alkan was witty, diabolical, fiercely misanthropic, and a true mystic (he was killed when a bookcase containing the Torah collapsed on him). Like all good Kabbalists, he had a wicked sense of humor, which, however, was tempered by a deep sense of melancholy and understanding of the human condition.

All of this is conveyed remarkably well by Lewenthal. Despite some tape saturation on some of the more violent passages, the power and delicacy of the piano are preserved. In other words, this is not your typical "digitized facsimile."

**7. Diamanda Galas, *Prayer and Malediction (Live)*, Asphodel/Mute, 53027 09841 (2LP).** Speaking of diabolical, I have saved the best for last. Few audiophiles have heard of dear Diamanda; however, I can't resist making a recommendation. Diamanda is a banshee: Hecate, Lilith, and Medusa all rolled into one—with a voice to match (you haven't heard low, until you've heard this).

But that's not all, Diamanda is not just a one-trick pony. This, her most accessible release, has her alone at the piano. The material is a blend of Negro-spirituals, cabaret, and lieder (with a little hellfire thrown in for good mea-

*(to page 95*